MiVoice Office 250

INSTALLATION AND ADMINISTRATION GUIDE

RELEASE 6.3 SP3

Mitel®

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Safety Instructions

For basic installation guidelines necessary for the proper and safe functioning of this equipment, refer to the *Important Safety Instructions* (document part number 835.3209) packaged with the system, included on the system software DVD, and posted on the eDocs Web site (http://edocs.mitel.com/).

NOTE Read and follow all information contained in the Important Safety Instructions document before attempting to install or use this product. Only trained, qualified service personnel shall install or maintain this product.

Regulatory Notices

For regulatory information for U.S., Canada, and Europe, refer to the Regulatory Notices section in the *Important Safety Instructions* (document part number 835.3209) packaged with the system, included on the system software DVD, and posted on the eDocs Web site (http://edocs.mitel.com/).

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For complete information on returning equipment, refer to the current *Mitel Repair and Return Policy*, document part number 835.1065. This document includes specific information on the following subjects: warranty, procedures to follow when returning equipment, equipment damaged in shipment, insurance, repair policy, and advance replacement policy.

Secure Socket Layer

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Network Security Statement

Although no telecommunications system or data network is entirely secure, as long as appropriate security measures are put in place and properly maintained by both the customer and the installing company, this Mitel Communications Platform architecture and its associated server-based applications are substantially secure against unauthorized access to the customer's data network via the telecommunications system. Appropriate security measures include, but are not limited to, the proper implementation of user/administrative accounts, passwords, firewalls, Network Address Translation (NAT), access control lists, virus protection, security updates, etc., and the proper maintenance of access points/programs and their respective accounts/passwords.

Chapter 1

MiVoice Office 250 New Features

Introduction

This section of the manual briefly describes the new features and enhancements available on the MiVoice Office 250.

Some of the items are documented mainly in this installation manual, some are documented mainly in the *MiVoice Office 250 Features and Programming Guide*, and some are documented in their respective application's documentation set or help files. For details on each item, refer to the indicated pages, document, or help file.

Important Introductions and Discontinuations

- The CS platform is no longer supported Release 6.2 and later.

NOTE CS controller hardware migration bundles are described in bulletins.

- Unified Communicator Express was discontinued in May 2015. It is now removed from MiVoice Office 250 Release 6.3 documentation.

MiVoice Office 250 v6.3 SP3 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the 6.3 SP3 software release.

Improved Functionality

Callback from Voice Mail feature can be used by SIP phones.

A Hunt Group Supervisor using a 6900 SIP Phone can now initiate Station Monitor, Barge-In and Steal. Barge-In and Steal features are available after activating Station Monitor on 6900 SIP Phone. This enhancement requires MiVoice Office Application Suite Release 5.2 or later.

Security Updates

New OpenSSL version 1.0.2r.

MiVoice Office 250 v6.3 SP2 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the 6.3 SP2 software release.

Improved functionality for SIP Phones, especially 6900 SIP Phones

- Intercom can be dialed using the keypad of the 6900 SIP phone (this improvement needs MiVoice Office Application Suite Release 5.1 SP2 or later).
- The default value of SIP devices for **Audio for Calls Camped onto this Device** is set to **Ringback** instead of **MOH (Music on Hold)**. When MOH is not installed, the user on-hold does not hear anything. Upgrading an existing system implements this new default setting.
- When adding off-node devices to Hunt Groups or Ring in Destinations, the device type IP/Digital is renamed as IP/Digital/SIP/6900 to make it evident that this needs to be selected only for 6900 SIP Phones.
- The default values for device type 6900 SIP phone are also suitable for the Phone Manager clients. It is now renamed as 6900/Phone Manager SIP phone.

Security Updates

- MiVoice Office 250 is now GDPR compliant.
- MiVO 250 offers TLS 1.2 on all secured system connections. The System Administration (Sys Ad) and the Database Programming (DBP) applications connect with TLS 1.2 to MiVO 250.
- The default value of OAI/WEB/ SIP ports are set to disabled. It is recommended to set this value to enabled before the ports are used. Upgrading an existing system implements this new default setting.

Support for Mitel 5613/5614 SIP phones

The Mitel 5613/5614 SIP Phones are supported on MiVoice Office 250 with the same functionality as the previous models of Mitel 5603/5604.

MiVoice Office 250 v6.3 SP1 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the 6.3 SP1 software release.

Improved OAI functionality to assist with 6900 SIP Phone Deployment

Single Provisioning: When a 6900 phone is created in MiVO250, it automatically generates authentication details. Using encryption, MiVO AppSuite can import the 6900 device authentication details. When configuration changes are made to 6900 SIP Phone Group(s) the MiVO250 will notify MiVO AppSuite via OAI.

Page zone numbers: With improved OAI functionality 6900 phones can now select a Page Zone ID in Keymap profiles on MiVO AppSuite.

NOTE If CT Gateway is in use, this will need to be upgraded to 5.0.63.0 to allow the new OAI functionality to be used.

Other Updates

AWP shows the 69xx SIP phone as separate device type in the Device General tab.

Security Updates

- New OpenSSL version 1.0.2n.
- **Meltdown/Spectre**: The new Bios on PS-1 Dell R230 servers is supported.
- GDPR documents are available as part of the Release 6.3 SP1 documentation.
- Inbound Authentication is now available for SIP Voicemail.
  When creating a SIP Voicemail, it is recommended to populate Inbound Authentication credentials to avoid any unauthorized access to MiVO250.

MiVoice Office 250 v6.3 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the 6.3 software release.

CloudLink with MiVoice Office 250

The CloudLink Gateway can be connected to the MiVoice Office 250 with SIP trunks. It allows MiVoice Office 250 Users (with DEE) to include their Mobile Device on which the **OfficeLink** App is installed. The **OfficeLink** App supports system features. For more description on **OfficeLink**, and how to configure **CloudLink** in MiVoice Office 250 DBP, see http://mitel.help.

No licenses are needed. The MiVoice Office 250 Release 6.3 detects CloudLink as a Trusted Application and automatically releases the needed OAI and SIP trunk licenses. The Trunk group configuration for CloudLink is supported with a new template for CloudLink.

Support of 6900 SIP phones with MiVO Application Suite 5.1

Mitel 6920, 6930 and 6940 SIP Phones are supported in conjunction with MiVO AppSuite 5.1.

The 6900 SIP phone configuration is supported with a new 6900 SIP phone type, that can be used instead of the generic SIP phone type.

The MiVoice Office 250 Rel 6.3 supports additional SIP extension and trunk features, handset flags and security improvements.

Administration Features, when using SIP devices

The MiVoice Office 250 R6.3 allows SIP devices to have Administrator and Door Relay rights. Clients on Application Suite 5.1 have special keys defined to activate: Door Relay, Night Ring on/off, Clear System Alarm, Diagnostics on/off, System History Freeze/Unfreeze, and Automatic Diagnostics on/off.

Divert/Accept UCD Calls, when using SIP devices

SIP devices support the Hunt Group Remove/Replace feature. So, the incoming hunt group calls to SIP phones can be stopped with the DBP flag, or by using the feature code (322 HG Remove, 323 HG Replace, 324 Toggle HG Remove/Replace).

SIP Reverse Transfer

Calls on hold or calls ringing on another phone or hunt group can be picked up by a SIP device.

Direct Page

Direct Page is an enhancement to the existing **Page** feature, which allows you to page to the loudspeaker of an individual IP/ Digital Phone instead of a whole Paging Zone.

- To configure an IP/ Digital Phone as a destination for Direct Page the phone must be assigned to a Paging Zone, which has the **Use zone for Direct Page** option enabled. This is disabled by default.
- A Direct Page is then activated by dialing **7** (Default Feature Code for Paging), followed by the configured **Paging Zone**, and then the extension number of the phone you want to Direct Page.
- Direct Page can only be activated by Phones that have the **Initiate Direct Page** flag set to **Yes**.
- Analog and SIP phones cannot receive a **Direct Page**, but can initiate a Direct Page to an IP/ Digital Phone.

Flexible SIP Header Configuration to SIP Trunk Provider

There are new fields available in the SIP Trunk Group configuration that allows to define which Caller ID information is sent to the SIP provider.

Restricted CLI in SIP Header

The caller identification can be withheld within the SIP header.

Display only Caller Identification instead of Transfer Information

A new Phone flag allows you to suppress transfer information and instead display caller identification and name when the incoming call is ringing.

Alternate Lamp Indication on Phone/ PKM Keys

Previously, MiVoice Office 250 lamp indication used three flashing rhythms (ringing: quick, hold from yourself: moderate, everything else: slow). A new system flag **Use alternate key lamp indication for Phone status** allows a clearer difference (ringing: slow, hold from anyone: moderate, active calls: continuously lit).

Release Key

Actual desktop phones on MiVoice Office 250 do not have the old “infinity key” to release a call with a simple keyclick. For example, a receptionist with a headset still had to use the hook in the cradle of the desktop phone to release calls. Now it is possible to configure the **release call functionality** on any phone key. The configuration can also be defined by the user in the User Web Portal (UWP).

Notification of no Primary or Local Attendant Configured

The Primary or local Attendant is important to system behavior. When a user unintentionally removes these from the Database configuration, a warning message appears.

SSL Certificate

You can generate **Certificate Signing Request** and **Upload SSL Certificate** settings to create a CSR request and upload customer’s own certificate on the system.

Security Updates

- New OpenSSL version 1.0.2l
- Improvements have been made to the existing Firewall, including the ability to save and restore settings. Firewall configuration options have been extended to include the PS-1 when equipped with the MiVoice Office 250 6.3 software release.

Discontinuation of UCX (Unified Communicator Express)

Unified Communicator Express was discontinued in May 2015. It is now removed from MiVoice Office 250 R6.3 documentation.

MiVoice Office 250 v6.2 SP2 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.2 SP2 software release.

UVM Diagnostic Improvements

The UVM diagnostic improvements supports changing logging level for the UVM application to help diagnose customer-affecting issues. The logging level for the UVM application can be only debug or info. The Debug logging level provides logs with the most information. This includes information that is useful for debugging configuration and hardware integration errors. The Info logging level includes information about changes in application state values and major application events. It is available in SysAd and in the AWP.

NOTE The debug logging level for UVM is only to be used for specific debugging activity. It is not recommended to enable debug logging level without prior consultation with Mitel Product Support. Alarm A150 `Extended APP Logging is Enabled` will appear until all applications are turned off debug logging level again. The Alarm 150 will not appear on administrator telephones, it will only appear within System Administration and Diagnostics, and Message Print.

Dirty COW Security Updates

Dirty COW (Dirty copy-on-write) is a computer security vulnerability for the Linux kernel that affects all Linux-based operating systems. It is a local privilege escalation bug that exploits a race condition in the implementation of the copy-on-write mechanism in the kernel's memory-management subsystem.

The MiVoice Office 250 v6.2 SP2 has included a series of fixes to prevent the Dirty COW vulnerabilities from affecting the system.

New PS1 Hardware Server

The R230 Server supports the MiVoice Office 250 PS1 software, and it is capable of running the 32-bit Debian 4.0 based distribution with the MiVoice Office 250 components.

Other Enhancements

- The MiVoice Office 250 Release 6.2 SP2 uses OpenSSH 7.1p1 and OpenSSL 1.0.2j.
- The MiVoice Office 250 Release 6.2 SP2 supports Dell R230 as PS-1.

MiVoice Office 250 v6.2 SP1 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.2 SP1 software release.

System Diagnostic Improvements

This feature supports “extended CP logging” to help diagnose customer-affecting issues, and new “archive options” for preserving various logs and statistics from the system. They are available in SysAdmin and in the Admin Web Portal (AWP).

NOTE The Extended CP Logging capability is only to be used for specific debugging activity. It is not recommended to enable these logs without prior consultation with Mitel Product Support. Alarm A150 ‘Extended CP Logging is Enabled’ will appear until all Extended CP Logging options are disabled again.

Watchdog improvement

The watchdog monitors activity of call processing. If call processing does not respond to the watchdog every 30 seconds, it will restart the MiVoice Office 250 system. New Logs are in the cp_watchdog_observer_log.txt file available in SysAdmin and Admin Web Portal (AWP).

T1/E1 Refresh

The FPGA component on the Dual T1/E1 card had to be replaced. The new card version is backward compatible to previous releases. Release 6.2 SP1 now also contains the firmware upgrade mechanism.

MiVoice Office 250 v6.2 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.2 software release.

SYSTEM PARK

System Park is a feature that allows calls to be parked by a phone, then either retrieved again by that phone, or retrieved by another phone.

- The Park/Pickup key with the same destination may be configured on several phones.
  - then any of these phones can park or retrieve calls from the same destination.
- Multiple calls may be parked on the same destination.
  - If in a call, the call will be added to the queue of parked calls.
  - If not in a call, the first parked call on this destination will be retrieved.
- Parked calls will recall the person who parked the call after a timeout
  - Phantom destinations use the new Park Recall Timer (default 180s, range 30-600s)
  - Hunt Groups use the existing Hunt Group Recall Timer (default 180s, range 1-65'535s)
- If a call is parked on a Park/Pickup key, the key will flash.
  - except if the Hunt Group has the ‘Group Call Pick-up’ flag disabled.
- Additional Notes:
  - Phones without a Park/Pickup key can use the features 'transfer' and 'reverse transfer' to park and retrieve calls from destinations like "Park Location 1". Therefore it is recommended to give simple numbers to such Phantom or Hunt Group destinations.
  - Assigning a Park/Pickup key to a Hunt Group destination is an enhancement to the existing Group Pick-up feature, allowing Group Pick-up using a single key press, whilst also allowing for a visual indication.

IMPORT AUDIO FILES FOR AUDIOTEX RECORDING ENHANCEMENT

Audiotex Recordings are custom recordings used by the MiVoice Office 250 Unified Voice Messaging (UVM) applications like the Auto Attendant (AA) and Call Routing Announcement (CRA) features.

Until now, Audiotex recordings could only be recorded using a phone with access to the Voice Mail System Administrator’s Mailbox. Each recording is associated with a recording number and assigned to the application(s) in Database Programming.

From release 6.2, you can now import audio files and use them for Audiotex applications, including Auto Attendant (AA) and Call Routing Announcement (CRA) features. This will improve the quality and fidelity of the recordings and allow pre-recorded, professional recordings for auto attendant greetings and announcements.

Recording requirements are as follows:

- At this time, only WAV files are supported.
- Recordings must be less than 5 minutes in length.

MIVOICE 5624 WIRELESS PHONE SUPPORT

The MiVoice 5624 Wireless Phone seamlessly blends the wireless needs of the user with a wide range of critical applications like voice messaging and integrates with fixed legacy applications within the organization.

The MiVoice 5624 Wireless Phone is Mitel's first VoWiFi (Voice over Wi-Fi) handset operating on 802.11n network; it supports the deployment of voice over a Wireless LAN (WLAN) without degrading the performance of an existing .11n network capacity. A MiVoice 5624 Wireless Phone-based solution delivers trouble-free WLAN vendor interoperability and scalability, as well as the capacity to integrate and communicate with a comprehensive range of external sources.

Notes:

- MiVoice 5624 Wireless Phone is already available outside North America.
- In North America the regulatory approval and introduction is in progress with MiVoice Business and will then also be available for MiVoice Office 250
- MiVoice 5624 Wireless Phone is Mitel's rebranding of the earlier certified Ascom i62 and is technically the same. (Mitel SIP Center of Excellence specifications, 14-4940-00310).

DISCONTINUATION OF SUPPORT FOR CS PLATFORM

With Release 6.2, CS platform is not supported.

OTHER ENHANCEMENTS

- Security improvements for SSL connections (Webserver, SSH) used on the system.
  - Open SSH 7.1
  - OpenSSL 1.0.2a
  - Web services are only accessible via TLS 1.2 encryption

More Features in Release 6.2

MOH (Music on Hold) for parked destinations

To change from Ringback to MOH for Parked Calls, two configurations are needed:

1. Change the 'Audio for Calls Ringing at this Device', from Ringback to Music, at the Park Phantom(s).
2. Change the 'Audio for Transfer to Ring' from 'Ringback', to 'Use Next Device's Audio Source', at the Trunk Group.

Group Pickup Enhancement

Assigning a Park/Pickup key to a Hunt Group destination is an enhancement to the existing Group Pick-up feature, allowing Group Pick-up using a single key press, whilst also allowing for a visual indication.

Other Enhancements

MiVoice Office 250 Release 6.2 supports Exchange 2016.

MiVoice Office 250 Release 6.1 SP1 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.1 SP1 software release.

Multi-Node Video Support

MiVoice Office 250 Release 6.1 SP1 supports video calls between SIP clients connected to different nodes. A SIP device connected to Node 1 can now make a video call to a SIP device connected to Node 2.This is achieved by using the Peer-to-Peer (P2P) connection between two SIP endpoints for the video calls for the endpoints connected to MiVoice Office 250.

The MiVoice Office 250 Release 6.1 SP1 supports the following features.

- Video calls between SIP clients connected to different nodes.
- Backward compatibility. A P2P video call from 6.1 SP1 node to 6.1/6.0/5.1 node is modified to an audio call.
- SIP video devices as remote extensions via MBG.
- Video calls across the network like any other desktop device.

SIP Phone Group – Password Alarm

MiVoice Office 250 Release 6.1 SP1 provides a new System Flag.

When enabled, Alarm 149 will be generated each time a SIP Phone registers with the MiVoice Office 250, where the password used for the registration is either the same as the SIP Phone extension number or it is left empty. The alarm is raised against the SIP phone group to which the SIP Phone belongs. The SIP Phone will still able to register and operate, however the alarm will notify the Administrator that security needs to be reviewed immediately for this SIP Phone Group. This alarm will be cleared automatically by the system as soon as the SIP Phone registers again using new credentials. This field is located in System\Flags.

Note: The Administrator must make the password secure and then clear this alarm.

This feature is enabled by default. Mitel recommends that you do not disable this feature. The Alarms Notification Feature within System Administration and Diagnostics is used to send an email to an administrator when a System Alarm is generated.

MiVoice Office 250 Release 6.1 New Features

This section briefly describes the new features, hardware, and enhancements available on the MiVoice Office 250 when equipped with the v6.1 software release.

Product Re-branding

Mitel has introduced new product naming as follows:

Previous Name	Current Name
Mitel 5000/5000 CP/Mitel 5000 Communications Platform	MiVoice Office 250
Mitel Database Programming	MiVoice Office 250 Database Programming.
Mitel System Administration and Diagnostics	MiVoice Office 250 System Administration and Diagnostics
Mitel 5000 Administrator Web Portal	MiVoice Office 250 Administrator Web Portal
Mitel 5000 User Web Portal	MiVoice Office 250 User Web Portal
Mitel Attendant Console	MiVoice Office 250 Attendant Console
Mitel Applications Suite (MAS)	MiCollab
Mitel Unified Communicator Advanced (UCA)	MiCollab Client
Mitel Border Gateway	MiVoice Border Gateway
NuPoint Unified Messaging	MiCollab Unified Messaging

NOTE For products related to releases prior to 6.1, naming will reference their previous, non re-branded names.

The re-branded interfaces for Database Programming, System Administration and Diagnostics, Administrator Web Portal, and the User Web Portal are updated with the new Mitel logo and color scheme. Additionally, the re-branding includes, but is not limited to, the following other MiVoice Office 250 interfaces:

- MiVoice Office 250 Installation Wizards
- MiVoice Office 250 Private Networking
- MiVoice Office 250 PS1 Server

New PS1 Hardware Server

The existing PS1 software supported hardware platform is limited to two server types the Dell R210 and Dell R220. Both server types are capable of running the 32-bit Debian 4.0 based distribution with the MiVoice Office 250 components.

As of May 2014, Dell retired the R210 Server and replaced it with the PowerEdge R220 Server.

MiVoice Office 250 PS1 software runs on the Dell R220 Server.

NOTE Debian is a Linux distribution that is composed primarily of free and open-source software, most of which is under the GNU General Public License, and packaged by a group of individuals known as the Debian project. The Debian Stable distribution is one of the most popular for personal computers and network servers, and has been used as a base for several other Linux distributions.

Virtualized PS1

The PS1 can now be deployed as a virtual appliance running in a VMware environment. Please see the MiVoice Office 250 Installation Manual for details regarding the installation and upgrade to virtualized PS1.

See About the PS-1 on **page 201** for additional details.

MiVoice Office 250 Attendant Console Refresh

Due to its historical heritage, earlier versions of the Attendant Console supported older operating systems such as Windows 98 and old drivers for HASP keys. These drivers do not support 64-bit versions of the Microsoft Windows.

As of 2014 new drivers for the Sentinel HASP are available from SafeNet Inc., and are compatible with a variety of 64-bit Windows versions, including Windows 7, Windows 8, and Windows 10.

The MiVoice Office 250 Attendant Console Release 3.5 can read data from valid HASP keys on the following Windows Operating systems:

- Windows 7 (32- and 64-bit)
- Windows 8 (32- and 64-bit)
- Windows 8.1 (32- and 64-bit)
- Windows 10 (32- and 64-bit)

Digital Telephone Refresh

The Digital Telephone refresh redesigns the existing 8568 and 8528 Digital Telephone hardware to eliminate the potential for obsolete parts with the new redesigned Digital Telephones.

No new features are added to the refreshed series of Digital Telephones. To the end-user there are no differences in the installation and operation between the refreshed Digital Telephones and the previous versions.

The refreshed MiVoice 8568 and 8528 Digital Telephones are supported on all Mitel 5000 CP / MiVoice Office 250 systems that currently support these phones.

Video Support for SIP to SIP Calls

MiVoice Office 250 Release 6.1 supports video capable SIP endpoints connected to the system when one SIP endpoint tries to establish a video call or escalate an established audio call to video. The peer-to-peer (P2P) functionality enables new SIP Peer endpoints to send and receive video packets directly to another IP device. It also allows support for Mid-Call escalation of SIP-to-SIP calls using signaling from voice calls to video.

In Database Programming, Peer-to-Peer Audio has been renamed Peer-To-Peer Media.

BRITISH TELECOM FSK CLIP CLID Enhancements

MiVoice Office 250 Release 6.1 allows the decoding of Calling Line Identification (CLID) information in countries which use variations of BT CLIP (British Telecom Calling Line Identification Presentation) signaling on analog trunks.

BT CLIP replaces ETSI DTMF Caller ID.

BT CLIP supports the delivery of Caller ID information on analog trunks in the UK, and the standard is commonly known as BT CLIP. BT CLIP is an on-hook capability that provides the user with information about the caller before actually answering a call.

A new trunk Service Type and timer parameters are added in MiVoice Office 250 Database Programming to support BT CLIP.

When configuring a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and British Telecom FSK.

SIP ACD Agent Login

SIP endpoints - including a SIP DECT, SIP Softphone, and a Mitel Phone Manager Softphone to login in to and logout from an ACD queue is supported.

SIP phones can be added to a Basic or UCD Hunt Group member list. Calls routed to a particular Hunt Group may be directed to a SIP phone.

For a SIP phone to use the ACD features (Log in, Log out, Wrap-Up Terminate), an OAI application must be used in conjunction with the SIP phone. Mitel Phone Manager is the recommended application.

The ACD feature codes cannot be dialled using a SIP phone.

ETSI DTMF Caller ID

MiVoice Office 250 no longer supports ETSI DTMF signaling for caller information.

When configuring a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and British Telecom FSK.

5000 CP v6.0 SP3 New Features

This section briefly describes the new features, hardware, and enhancements available on the Mitel 5000 CP when equipped with the v6.0 SP3 software release.

There are no impacts to any of the 5000 CP documents or help files as a result of the following new features in release 6.0 SP3.

Shellshock Security Updates

Shellshock is a family of security bugs in the widely used Unix Bash shell. The Shellshock vulnerabilities affect various Unix-based systems and are used to execute command lines and command scripts.

The 5000 CP v6.0 SP3 has included a series of fixes to prevent the Shellshock vulnerabilities from affecting the system.

New 6.2.0.6 IP phone Firmware

5000 CP v6.0 SP3 introduces a new IP firmware version 6.2.0.6. The 53xx series IP Phones will be upgraded to this version once the 5000 CP v6.0 SP3 is installed.

Mitel SIP DECT 6.0 support

In 5000 CP v6.0 SP3, the SIP DECT Version 6 offering is fully supported and validated.

Microsoft Exchange 2013 support for Unified Voice Mail

In 5000 CP v6.0 SP3, Microsoft Exchange 2013 is fully supported when using it in conjunction with the Unified Voice Mail integration.

5000 CP v6.0 SP2 New Features and Enhancements

DTMF Decoding Payload

SIP Peers in Database Programming can be configured with the necessary custom payload type value, depending on the supported payload type of the far end SIP provider. In previous releases, this value was configured in IP Settings. In Release 6.0 SP2 and up, the DTMF Decoding Payload is now located in the individual configurations for SIP Phone Groups, SIP Trunk Groups, and SIP Voice Mails. The payload type is typically negotiated between carriers and devices, but in some cases the SIP service provider does not accept requests. The DTMF Payload Type is made available by the SIP trunk service provider.

Database Programming now allows the DTMF Payload type to be selected for SIP Peers, from a range of values that correspond to those supported by the SIP service provider.

The default DTMF Decoding Payload value applies to North American, UK, and Australian systems.

Propagation of Original Caller ID to P-Asserted Identity

In Database Programming, a new flag, *Do Not Propagate Original Caller ID to P-Asserted Identity*, was added. The administrator can set this flag when the Dynamic Extension Express (DEE) feature is used, and a DEE destination is reached via a SIP trunk and the *Propagate Original Caller ID* flag is enabled.

Do not use Out-bound Username in REGISTER

In Database Programming, a new flag, *Do not use Out-bound Username in REGISTER*, was added. This flag controls the registration of a SIP call when upgrading to Release 6.0 SP2.

If the flag is set to No, the to/from/contact headers in the SIP register correspond to the out-bound username. If the flag is set to Yes, the to/from/contact headers in SIP register correspond to the trunk group Caller ID.

Support for new DDM16 and DEM16 circuit board revisions

New hardware revisions of the DDM16 and DEM16 circuit boards are fully supported by the Release 6.0 SP2 software.

MCD Interoperability Support

The Mitel 5000 Release 6.0 SP2 maintains all current support with the published MCD Interoperability.

Microsoft Lync Integration Support

The Mitel 5000 Release 6.0 SP2 now has support for Microsoft Lync via the Micollab 6.0 product. Refer to the MiCollab 6.0 documentation for details on this integration.

Support for New Devices

The Mitel 5000 Release 6.0 SP2 supports the following new devices:

- The 5610 IP Dect Phone is now supported via a Media Border Gateway connection to the Mitel 5000 CP.
- New models of the 5320e phone have a backlit display. The Mitel 5000 v6.0 SP2 contains new firmware for the 5320e IP phones.
- Apple IOS support via a Media Border Gateway connection to the Mitel 5000 CP. Refer to the Media Border Gateway and IOS documentation for details.
- Mitel Phone Manager (formerly by Xarios).

UPS Monitoring Support Changes on 6.0 SP2 and later software

Effective Release 6.0 SP2, monitoring capabilities for previously approved suppliers of UPS units cannot be guaranteed to be compatible with the UPS Monitoring feature.

See “Compatible UPS Units” on **page 63**.

5000 CP v6.0 SP1 New Features and Enhancements

Mitel 5000 CP Engineering Guidelines

Formerly an appendix to this manual, the engineering guidelines for the 5000 CP have been created in a stand-alone document. See the *Mitel 5000 CP Engineering Guidelines* for more information.

Mid Call Features

Mid-Call Features provide a way for a mobile user to perform hold, transfer, and conferencing features when the 5000 places a call via Dynamic Extension Express (DEE).

NOTE Mid-Call Features are only supported with other networked 5000 CPs running Release v6.0 SP1 and higher software.

MBG SIP Services

The MBG (Mitel Border Gateway) is a platform for the secure deployment of multiple network connectivity services in a number of network edge scenarios. New MGB SIP features include:

- SIP device/trunk support
- UCA application support

Adding Local Dialing Prefixes

For Incoming calls on SIP trunks in the UK and Australian markets, where the leading prefix was not presented, the Mitel 5000 will now automatically add the following dialing prefixes:

- 0 for national calls in the UK and Australian markets
- 00 for international calls in the UK market
- 0011 for international calls in the Australian market

This will allow the called party to redial the external number back successfully, and it will allow incoming calls to match System Speed Dial numbers.

Database Programming Updates

The following updates to Database Programming have been implemented to improve usability as more routines become automated requiring less input from the administrator:

- Configuration Wizard suggesting Voice Mail applications types.
- Configuration Wizard Summary page lists Voice Mail applications programmed from within the Wizard, and those that have not been programmed.
- Phone users have the ability to access the Administrative Web Session via User Login/Password.
- Create an unassociated user.

After upgrading to Release 6.0 SP1, only the System Administration and Diagnostics (SysAd) shortcut displays in the Windows Start menu.

5000 CP v6.0 New Features and Enhancements

New features and enhancements introduced in release 6.0 software release are described in the following sections.

Axxess Networking Support

Further to product bulletins released during the 5.1 release cycle, Axxess networking is no longer supported beginning with the v6.0 release of the 5000 CP.

Configuration Wizard

The Configuration Wizard has been updated to make start-up easier, and to easily configure a variety of system level parameters including IP and e-mail settings, as well as numbering plans, baseline extensions, expansion modules, IP phones, voice processor options, and a variety of features and devices.

User Creation Wizard

The User Creation Wizard allows an administrator to create and edit individual users and their corresponding features and extensions from one area.

Database Programming Enhancements

Database Programming has been enhanced in a number of areas. The items below reflect the areas affected.

- Administrative Accounts
- Associated Extensions
- Programming Trunk Group Options
- Audio for Calls Holding for a Device
- Configuring SIP Peer Programming Options:
  - Support Menu Softkeys
  - Disabling Domain Validation
  - Display Updates
  - Ad Hoc Conferencing
- User Programming Fields
- User-Related Information
- Dynamic Extension Express
- Ring Duration
- Programming Dynamic Extension Express
- Area Flags
- Importing User Entries from CSV Files

- Device Baseline Extensions

DHCP Server

The 5000 CP now has an embedded DHCP server. The embedded DHCP server makes the 5000 CP capable to automatically provide the proper DHCP Options for programming IP phones, thereby reducing the chances of misconfiguration, and reducing installation time.

Bundled Save and Restore

The Database Operations menu is renamed Backup Operations. Two new options for database backup save and restore are located here. The Default Database option is now under the main Operations menu.

The ability to encrypt sensitive data is also provided.

The DB Test and Repair and DB Converter user interfaces are redesigned and incorporated into the new Backup Utility window. Scheduled backups are updated to allow for a bundled save.

Connection Wizard

If the Configuration Wizard is not displayed at startup, a new dialog listing connection tips for the Processing Server to connect with the Base Server will display instead of the Connection Wizard. This dialog will only be displayed when connecting to a Processing Server in online mode that is disconnected from the Base Server.

SIP Peer to Peer

A SIP Peer can now be put into a Network Group that supports peer-to-peer audio. The Peer-to-Peer (P2P) functionality enables the new SIP Peer devices to send and receive audio packets directly to the other IP device (without backplane connections). This implementation is the same as the existing 5000 IP devices.

Support for ETSI DTMF Caller ID

Caller ID information is decoded in countries which use varieties of ETSI DTMF signaling for caller information.

When configuring a Loop Start trunk, the Service Type options now include Bellcore FSK Caller ID and ETSI DTMF Caller ID.

Administrator Web Portal

The Administrator Web Portal (AWP) interface for the 5000 CP replaces the Administrative Web Session (AWS) as of Release 6.0. The AWP maintains the functionality of the AWS, and also allows simple adds, moves, and changes for users and phones. See the AWP application for details.

Mitel Border Gateway (MBG) Support

53xx series IP phones for Teleworker operation are now supported when an MBG is connected to a Mitel 5000 CP. See “Mitel Border Gateway (MBG) Connectivity” on **page 44** for details.

Support for new SIP Phones

Mitel 5603/5604/5607 SIP phones are now supported. Ad-Hoc Conferencing using these new SIP phones is also supported.

Alarms/Alerts via E-mail

This new feature in SysAd adds the ability to program user notification for alarms. Users are created and managed in DB Programming, as is their notification destination (e-mail, text message, etc.). In SysAd the alarms can be enabled or disabled of notification on a per user and individual alarm and/or alarm group basis.

Mitel UC 360 Collaboration Point Support

The 5000 CP supports the Mitel UC360 Collaboration Point, and it is programmed as a SIP device. Details are included in the *MiVoice Office 250 Features and Programming Guide*.

UVM E-Mail Sync with Gmail and Office 365

The UVM application provides integration and synchronization capabilities between an user’s voice mailbox and their e-mail client application and is referred to as E-Mail Synchronization. It currently has the ability to synchronize with e-mail using IMAP with Exchange, Lotus Notes and Groupwise.

With the popularity of hosted e-mail providers, the purpose of this enhancement is to extend the capabilities to include Gmail and Office 365.

See the *MItel 5000 Communications Platform Unified Voice Messaging E-Mail Sychronization Adminitrator Guide* for details.

DTMF Payload Type 101

DTMF payload type 101 (one of the most common and preferred standards in SIP calls) is now supported.

Combination of Inactivity Timers

System Administration & Diagnostics and Database Programming now use the same timer for inactivity. The timer is programmed in the Options window on the Advanced Tab in System Administration & Diagnostics.

Terminology Changes

To accommodate changes in technology and/or to better align with the overall corporate marketing strategy, the following product terminology changes have been implemented in the v6.0 release:

- The Database Operations menu is renamed Backup Operations
- System Administrative Accounts is renamed Maintenance Accounts
- The Administrative Web Session (AWS)is renamed the Administrative Web Portal (AWP)

Chapter 2

Document Overview

Introduction

This manual provides the information needed to plan, install, perform initial programming, implement, and maintain the MiVo250Communications Platform (CP).

This manual provides information for new installations, upgrades, maintenance and troubleshooting relating to installation starting from release v6.0. For prior releases, refer to the documentation provided for that release.

All other system information is in the *MiVoice Office 250 Features and Programming Guide* (part number 580.8006), which covers all features, programming, diagnostics and troubleshooting, and any other information other than installation-related. You can find all documentation online at the Mitel eDocs Web site (http://edocs.mitel.com/).

Intended Audience

NOTICE

To receive technical assistance from the Mitel Technical Support group, you must have the following certifications:

- MiVoice Office 250 Basic Technical Certification.
- Telecommunications Industry Association (TIA) Convergence Technologies Professional (CTP) certification.

If you do not have these certifications, Mitel Technical Support will provide assistance, but one Technical Support Incident (TSI) will be charged.

This manual has been developed for Mitel-trained technicians who:

- Possess the knowledge and skill required to set up, maintain, and troubleshoot a programmable circuit-switched telephone system.
- Use standardized telecommunication practices, procedures, and troubleshooting methods when performing their work.
- Possess the ability to effectively apply the principles of:
  - Internet Protocol (IP) telephony/computer telephony (CT).
  - Voice over IP (VoIP), including Quality of Service (QoS).
  - Data networking.
  - Technology convergence.

Chapters in This Manual

This manual includes the following chapters:

- **MiVoice Office 250 New Features**. Provides an overview of the new hardware, features, and enhancements in this release.
- **Document Overview**: This chapter (the chapter your are currently reading) provides information about the document's structure and content, associated documents, and typographical methods used to call the reader's attention to important information.

- **Product Description**: Describes system processes, feature summaries, and licensing requirements.
- **Specifications**: Defines capacities, dimensions, and parameters of the hardware and software components of the system. The chapter also identifies the environmental conditions required to ensure safe and reliable system operation.
- **Installation**: Describes the tasks and procedures required to correctly install, upgrade, and maintain the system, digital interface equipment, phones, and optional system components.
- **Reference**: Provides ancillary information, including:
  - Mitel Technical Support contact information.
  - Abbreviations and acronyms used in the manual.
  - Parts ordering guidelines.
  - Lists of selected hardware, software, license, and documentation part numbers.
- **Appendixes**: The manual also provides appendixes that contain detailed information about related technical topics.
  - **Private Networking**
  - **Network IP Topology**
  - **Open Source License Agreements**
  - **Phones**

Mitel Reference Documents

The following documents provide detailed information for installing, programming, and using the MiVoice Office 250 and peripheral equipment:

MiVoice Office 250 Documents

- *MiVoice Office 250 Installation Manual*, part number 580.8000
- *MiVoice Office 250 Features and Programming Guide*, part number 580.8006
- *Mitel System Administration & Diagnostics Guide*, part number 550.8125
- *MiVoice Office 250 Phone Administrator Guide*, part number 580.8001
- *MiVoice Office 250 Unified Voice Messaging Administrator Guide*, part number 580.8009
- *MiVoice Office 250 Installation Quick Start Guide*, part number 835.2879
- *MiVoice Office 250 Program Planning Sheets*, part number 835.2882
- *MiVoice Office 250 Memory Card Replacement Instructions*, part number 835.3033

NOTE For v4.0 and later, all of the information contained in the *MiVoice Office 250 Reference Manual* has been incorporated into this installation manual and the *MiVoice Office 250 Features and Programming Guide*. The Reference Manual is no longer needed or available with v4.0 or later systems.

Peripheral Equipment Documents

- *CT Gateway Installation Manual*, part number 835.2161

- *IP Device Power Supply Compatibility Matrix,* part number 835.2688
- *MiVoice Office 250 and NuPoint UM Integration Guide*, part number 580.8008
- *Message Print Diagnostics Manual*, part number 550.8018
- *System Equipment Grounding Guidelines*, part number 835.2845
- *System OAI Toolkit Specifications Manual*, part number 835.2615
- *Universal RS-232 Kit Instructions*, part number 828.1282
- *Voice Processing Diagnostics Manual*, part number 550.8019
- *VoIP Data Network Requirements*, part number 835.2885
- *VoIP Design Considerations, Guidelines, & Practices*, part number 835.2976
- Titles of applicable phone user guides appear in the Reference chapter, Table 69 on **page 378**.

Document Style Conventions

This section identifies the style conventions used throughout the manual. An extensive list of abbreviations and acronyms used in this manual appears in the Reference chapter. See “Abbreviations and Acronyms” on **page 1-380**.

British English and American English References

This document supports the requirements of technicians in the United States of America (USA), Canada, and Mitel European market, which includes the United Kingdom (UK). Because of this dual North American and European support, dual references are made to industry features, standards, and jargon, as appropriate throughout the document. Terms used in the Mitel European market appear in brackets [<*European equivalent*>]. For the purposes of this addendum, British English terms are assumed to apply to other English-speaking European cultures, as well.

For example, the U.S. telecom industry refers to an audio communication line between a public switching system and a private switching system as a “central office (CO) trunk.” In the UK, this same type of line is called a “local exchange trunk.” Applying the dual-reference guideline would result in the following statements:

- During system installation, the technician connects CO trunks [local exchange trunks] to the LSM-2 or LSM-4 modules.
- To turn the Diagnostics Mode on or off at an administrator phone, the system administrator enters 9900 [9100 in Europe].

However, this document does not make a similar distinction between American English and British English spellings of common words. Only American English spellings appear in this document. For example, the word “analog” is *not* also spelled “analogue,” inasmuch as the meaning is clear.

References to “Endpoints”

Starting with the v4.0 release:

- Digital endpoints are now referred to as digital telephones.
- IP endpoints are now referred to as IP phones.

Prior to v4.0, this document used the term “endpoint” to describe the devices that the MiVoice Office 250 family of products supports over Internet Protocol. In this context, digital and IP phones are endpoints, as are data modules, fax machines, computer telephony (CT) gateways, Single-Line Adapters (SLA), IP Single-Line Adapters (IP-SLA), Media Gateway Control Protocol (MGCP)/Session Initiation Protocol (SIP) gateways, Audio and Web Conferencing (AWC), and answering machines.

To promote clear communication and understanding, commonly accepted names of endpoints are used as appropriate. In particular, desktop IP phones may also be referred to as IP phones. Digital endpoints may also referred to as digital telephones. Wireless endpoints may also referred to as wireless phones. And, analog single line endpoints are also referred to as single line phones or single line sets.

NOTE In this document and in the Database (DB) Programming application, an endpoint or phone may also be called a “station” or “extension” if reference to the endpoint or phone and its environment are intended. Or, it may be called “device” in the most general sense.

Hazard Identification, Notices, and Notes

This manual uses standardized graphic conventions to promote safe practices by technicians and to prevent damage to equipment and data. Potentially harmful situations are identified according to the degree of hazard the technician is exposed to. Relevant policies appear in a boxed format, while important information and notes are set apart from the text with attention-getting labels. This section explains the conventions used and their meanings.

Hazard Identification Methods

The following hazard identification methods appear in this manual.

- A WARNING ⚠ identifies a hazard that, if not heeded, could cause definite personal injury.

WARNING

Electrical Shock. Disconnect the computer from all power sources before opening the cover and working inside the chassis. Failure to take this precaution could result in electrical shock and injury.

- A CAUTION ⚠ identifies a hazard that, if not heeded, could cause minor personal injury.

CAUTION

Pinch Hazard. Keep fingers clear of the edges of the faceplate when inserting a trunk module into the chassis.

- A NOTICE identifies a hazard that, if not heeded, could cause damage to equipment or corruption of data or databases. Notices also stipulate conditions, requirements, and/or policies relevant to the topic under discussion.

NOTICE

Equipment Damage Hazard. Only use a single appropriate power adapter. Do not connect a U.S. power supply and a Universal power supply to the same device. Also, when using the Universal Power Supply with a barrel connector, no devices (for example, hubs) should be inserted between the KS/SLA jack on the adapter and the LAN jack on the phone because power is supplied through the cable.

Other Notes

Notes and Important messages provide supplemental information about the topic being discussed.

- An IMPORTANT message contains information that requires special attention related to the topic.

IMPORTANT The chassis must connect firmly to earth ground at the Ground Lug.

- A NOTE conveys supplemental information related to the topic.

NOTE Commercially available premoistened towels provide adequate cleaning.

Chapter 3

Product Description

Chapter Overview

This chapter discusses the MiVoice Office 250 family of products that provide an enhanced IP solution for small- and medium-sized business environments. The MiVoice Office 250 is a family of converged communications platforms that supports Internet Protocol (IP), digital, and analog single line phones.

System Description Changes

Prior to v4.0, the system was marketed as the CS-5200, CS-5400, and CS-5600 communication servers. Starting with the v4.0 release, these CS-5200, CS-5400, and CS-5600 product model designations are no longer used, and the original CS-5x00 hardware platform is now referred to as the CS Controller platform. The newer hardware platform is referred to as the HX Controller platform.

Also starting with the v4.0 software release, hardware expansion options have been simplified and are now independent of licensing. For example, a new license is no longer required to add a Processor Expansion Card (PEC-1) or a Processing Server (PS-1) to the system. Release 6.1 introduced the PS-1 as a virtual appliance. As a result, system capacities that were tied to the 5x00 size designations (e.g., a CS-5200 being limited to 75 IP phones) are no longer enforced. A system can now be equipped with up to 250 phones (up to 240 digital or up to 250 IP) as long as the phones are properly licensed.

NOTE Starting with the v4.0 software release, the lack of a PEC-1 and/or a PS-1 (or vPS-1) no longer reduces the maximum feature capacities available on the system. For a list of capacities, see “System Capacities” on **page 1-67**.

Instead of using Digital Expansion Interface (DEI) licensing, digital telephones are now licensed the same way that IP phones are licensed in the system. A new category E license was added in v4.0 for digital telephones. When upgrading existing systems to v4.0 or later, the AMC will convert the old DEI licensing model to the category E licensing model. Each DEI license is converted to 48 category E licenses. Note that SLAs attached to digital circuits do not require a license.

Due to these changes in licensing, the IP Endpoints Enabled and System IP Endpoint Capacity values have been eliminated from the Software License Feature folder in DB Programming. Also, an Extended Node Capacity license is no longer required to go above 63 nodes.

System Evolution

Starting with v2.0, G.711 became the default vocoder for MiVoice Office 250 instead of G.729. This change provided increased IP phone capacity by using a less processing-intensive codec to allocate IP resources than was used in preceding versions.

Also since v2.0, the system has supported four single line phones with the Single Line Module (SLM-4), part number 580.2100. Prior to the SLM-4, single line phones were supported on the MiVoice Office 250 by digital Single-Line Adapters (SLA), IP SLAs, or the two single line ports built into the Base Server. The built-in single line ports are wired to connect with two built-in loop start trunk ports, which provides the system with power fail bypass capability.

In addition, with v2.1 or later, any system equipped with one or two Digital Expansion Interface (DEI) chassis can run up to three eight-port Single Line Modules (SLM-8), part number 580.2101, in each DEI. Having the same form-factor as the Digital Endpoint Module (DEM-16), the SLM-8 fits in the DEI chassis.

For systems that utilize numerous IP phones, IP trunks, and IP networking, v2.0 and later allows oversubscription of IP resources. With oversubscription, dynamic allocation of IP resources, and Camp On capability, a system can be programmed with more phones than it has IP resources available to support simultaneously. For oversubscribed systems, the Resource Reservation Tool allows you to reserve IP resources for essential functions such as Emergency/911, as well as other customer-specified devices and phones.

MiVoice Office 250

This section summarizes the technology of the newer MiVoice Office 250 HX Controller and original CS Controller Base Server and identifies system interfaces to network and customer environments.

MiVoice Office 250 Technology

Built on 32-bit architecture and packaged in a 2U chassis (HX Controller) or 1U chassis (CS Controller), the MiVoice Office 250 provides a robust IP communication solution. With appropriate licensing, the system delivers communication assets that include the following services and features:

- Linux® operating system
- Loop start, T1, T1/Primary Rate Interface (PRI), E1/PRI, Basic Rate Interface [European market only], and IP gateway (MGCP and SIP) trunk connectivity
- Messaging, including Unified Voice Messaging and E-Mail Synchronization
- Networking over IP
- Networking over T1/E1 spans
- Fax over IP (FoIP)—T.38—MiVoice Office 250 to MiVoice Office 250 nodes only
- Compact flash-type memory card for storing system applications, customer database, and voice mail messages
- USB flash drive for backing up voice mail
- Support for IP, SIP, Multi-Protocol, Wireless IP, Multimedia Touch Screen, digital, and analog single line phones
- Programmable Call Processing features for enhanced business productivity
- Directory Look-Up
- Hunt Groups
- Automatic and Uniform Call Distribution
- Open Architecture Interface (OAI)
- Multilingual Support
- Music-On-Hold
- Paging
- Power Fail Bypass circuitry
- Wide Area Network (WAN) Fail Survivability

Unified Voice Messaging (previously called Basic Voice Mail) service is provided with the system, and additional voice mail capacity is available by installing the NuPoint Unified Messaging (UM) system. For details, refer to the *MiVoice Office 250 and NuPoint UM Integration Guide* (part number 580.8008).

Digital signal processor (DSP) technology enables Conferencing, Dual Tone Multi-Frequency (DTMF) detection, tone generation, modem operation, and Automatic Number Identification (ANI), or Caller ID.

The MiVoice Office 250 can operate as a stand-alone system or as a node in a network of up to 99 MiVoice Office 250 nodes.

MiVoice Office 250 Interfaces

The MiVoice Office 250 provides multiple access ports to the features and functionality of the platform. From the front of the chassis, the technician can set-up the system and program it if desired, read system alarm messages, backup voice mail messages, and monitor ongoing system events. From the back of the chassis, the technician can connect system power, a local area network (LAN), trunks and spans, two single line devices, two loop start trunks, and an external music source, paging system, and voice mail system. See “Specifications“ for details.

Optional Processor Expansion Card

Attaching an optional Processor Expansion Card (PEC-1) to the system’s Processor Module (PM-1) provides additional IP and/or voice processing resources. See the MiVoice Office 250 Enginerring Guidelines for details on when a PEC-1 might be needed.

Optional Processing Server

An optional Processing Server, available as both a physical device (PS-1) and virtual applicane (vPS-1) can be networked to the system to provide the additional call processing power needed for many larger installations. See the MiVoice Office 250 Enginerring Guidelines for details on when a PS-1/vPS-1 might be needed.

With v2.1 or later, the PS-1 supports 16-port Unified Voice Messaging (previously called Basic Voice Mail) and handles call processing chores for the entire system. It controls and communicates with the Base Server across the customer’s local area network (LAN). For rlease 5.0 and higher, UVM can now be licensed for up to 24 or 32 voice mail ports, as follows:

- 24 UVM ports (maximum) on a standalone CS Controller (without a PS-1/vPS-1)
- 32 UVM ports (maximum) on a standalone HX Controller (without a PS-1/vPS-1)
- 32 UVM ports (maximum) on a CS or HX Controller equipped with a PS-1/vPS-1

For systems that utilize numerous IP phones, IP trunks, and IP networking, v2.0 and later allow over-subscription of IP resources. With over-subscription, dynamic allocation of IP resources, and Camp On capability, a system can be programmed with more phones than it has IP resources available to support simultaneously. For oversubscribed systems, the Resource Reservation Tool allows you to reserve IP resources for essential functions such as Emergency/911, as well as other customer-specified devices and phones.

Digital Capability

The MiVoice Office 250 HX Controller platform has Integrated support for up to 48 digital desktop devices (without the need for a secondary digital expansion chassis). If more than 48 digital devices are required, up to four optional Digital Expansion Interface (DEI) units can be attached to the system.

Although the original CS Controller platform does not have native digital support, it too can be equipped with up to four optional DEIs.

Digital Desktop Module

Each Digital Desktop Module (DDM-16) provides 16 circuits for connecting a variety of digital telephones. There can be up to three DDM-16s installed in any of the four HX Controller expansion bays. Each DDM-16 includes:

- A single 48-pin connector and a special DDM cable for connection to a station block at the Main Distribution Frame
- Status LEDs (online, busy, and remove)
- Module release lever for safely removing the module from the bay
- DSP resources (4 DTMF receivers, 5 tone generators, and 16 speakerphones for devices connected to the module)

Note that the new DDM-16s are not subject to any of the voice channel trade-offs that the original DEM-16s are subject to when used in two-DEI or four-DEI mode.

Digital Endpoint Module

Each Digital Endpoint Module (DEM-16) provides 16 circuits for connecting a variety of digital telephones (including 8500, 8520, 8528, 8560, 8568, Executive, Professional, Standard, Associate, and Basic) or Single-Line Adapters (SLAs) to the system.

Each DEM-16 is also equipped with a single Digital Signal Processor (DSP) to provide the following module and/or system resources:

- **Speakerphone resources** for all digital telephones connected to the module.
- **Tone generating circuit** for providing system tones (busy, reorder, Do-Not-Disturb, etc.) to the phones connected to the module, if necessary. It is available in case there are no voice channels available in the system to issue tones.
- **Switching matrix** to allow access to any available time slot on the module's PCM highway (bus). This is provided since the module may need up to 32 simultaneous voice channel connections.

Single-Line Adapter

A Single-Line Adapter (SLA), which can be installed on an available DDM-16 or DEM-16 circuit, is used for connecting two AC-ringing, single line DTMF sets, playback devices, and/or off-premises extensions.

With a system installed in the United States, an SLA can also be used for connecting two Direct Inward Dialing (DID) trunks—or, one single line phone and one DID trunk. For more information on using DID trunks, refer to the *MiVoice Office 250 Features and Programming Guide*. SLAs are shipped with both ports set for single line device operation. Before connecting a DID trunk, be sure to open the unit and set the corresponding port for DID operation. Also, if necessary, adjust the hybrid balance setting to better accommodate longer loop lengths. For installation instructions, see the **Installation** chapter.

AC-ringing off-premises extensions can be connected directly to the SLA circuit. E*xternal* high-voltage ringing adapter (HVRA) units, power supplies, and ring generators are *not* needed.

IMPORTANT SLAs do not support *visual* Message Waiting applications. If single line sets with Message Waiting lamps are installed on SLAs, the lamps will not function. However, single line sets can be configured in DB Programming to receive *audible* Message Waiting tones.

Each SLA, whether used for single line sets or DID trunks or both, is powered by its own AC transformer. Also, SLAs have two self-correcting thermistors. If a short circuit causes a thermistor to open, it will automatically close when the short is removed.

NOTICE

Possible system degradation. For optimal system performance, do not use SLA circuits for Voice Mail and Automated Attendant functions. Rather, the Voice Processing PC and applications software should be used for these purposes.

Data Port Modules

The Personal Computer Data Port Module (PCDPM) and Modem Data Port Module (MDPM) are connected to appropriate digital telephones to provide hard-wire paths necessary to connect compatible Programmable Key Modules (PKMs) or Digital Station Select (DSS)/Busy Lamp Field (BLF) units. Contrary to their names, however, neither device supports serial data.

Personal Computer Data Port Module

The PCDPM can be installed on Standard, Associate, Executive, and Professional Display phones or on 8520 or 8560 phones to gain Off-Hook Voice Announce (OVHA) receive capability or to install an optional Modem Data Port Module (MDPM).

NOTE The PCDPM supports audio only when used in conjunction with the 5000 DEI product. In this version of software, the serial port is not functional on the PCDPM, and it does not transmit data, even though the name of the device suggests that it does.

Modem Data Port Module

The optional Modem Data Port Module (MDPM) can be installed on Standard Display, Executive Display, 8520, or 8560 phones for attaching a modem-equipped data device or a single line phone. The MDPM requires a PCDPM.

The Mitel Programmable Key Module (PKM) 16 (previously called the Model 8416 Mini-DSS) connects to Mitel 8520, 8560, or 8660 phones without PCDPM and MDPM devices. The only programming required for the PKM 16 (Mini-DSS) is a keymap, which is explained in detail in the *MiVoice Office 250 Features and Programming Guide*.

Trunks and Phones

The MiVoice Office 250 supports the following trunks, spans, gateways, and phones:

- **Central Office (CO) [Local Exchange] Trunks**: Up to 12 (CS Controller) or 16 (HX Controller) loop start [local exchange] trunks can be installed in increments of four trunks per LSM-4 module. In addition, two (CS) or four (HX) loop start trunks can be installed in the RJ-14 port built in on the Base Server.
- **Spans**: Up to six (CS Controller) or eight (HX Controller) T1, T1/PRI, or E1/PRI spans can be installe:.
  - *T1*: In Database (DB) Programming each of the 24 channels of a channelized T1 span can be enabled as any of the following types of channels or circuits:
    - Loop Start
    - Ground Start
    - DID (Direct Inward Dial)/[in Europe, DDI (Direct Dial Inward)]
    - E&M (Ear and Mouth/2-way)
    - OPX (Off Premises Extension)
    - *T1/PRI*: A T1/PRI span provides 23 information-bearing (B) channels and one data (D) channel for signaling and control.
    - *E1/PRI*: An E1/PRI span provides 30 B channels and one D channel.
- **IP Devices**: The MiVoice Office 250 supports IP phones, digital telephones, and IP Single-Line Adapter (IP-SLA) devices. The Internet Protocol Resource Application (IPRA) supplies IP functionality for the system and communicates to the phones and gateways across 100Base-T Ethernet. The IPRA does ***not*** support legacy IP SoftPhones. See page 376 for specific phone models.
- **8602 Softphone Application**: 8602 application software is distributed separately from the system software. Performance and interoperability is equivalent to hard IP phones. For details, refer to the *Model 8602 User Guide*, part number 835.3019.
- **IP Gateway Trunks**: The MiVoice Office 250 supports Media Gateway Control Protocol (MGCP) and Session Initiation Protocol (SIP) gateway trunks.
- **Programmable Key Module (PKM)**: Previously referred to as **Direct Station Select (DSS)/Busy Lamp Field (BLF)**. The MiVoice Office 250 supports a variety of PKMs, including the PKM 12, PKM 48, PKM 50, and PKM 60. For details, see “Direct Station Select/Busy Lamp Field Units” on **page 1-116**.
- **PKM 16**: The MiVoice Office 250 supports a Mitel Programmable Key Module (PKM) 16 (previously called the Model 8416 Mini-DSS) that provides one-button access to 16 extension numbers and/or feature buttons/resources. For details, see “8416 Mini-Direct Station Select (Mini-DSS) Unit” on **page 1-117**.
- **Wireless Phones**: 8664, 8665, and 8668 are wireless IP phones. For details, refer to the *NetLink SVP Server and Model 8664/8665/8668 Installation Manual*, part number 935.4521, and/or to the *Model 8664/8665/8668 User Guide*, part number 935.4522.

- **Single Line Phones**: In addition to the two circuits manufactured, single line phones can be installed with IP-SLA devices, Single-Line Adapters, and four-port and eight-port Single Line Modules (SLM-4s and -8s).

Mitel Border Gateway (MBG) Connectivity

The MiVoice Office 250 release 6.0 supports connectivity to the MBG. For teleworker support, the following MiNET phones are supported:

- 5304 (supported only with firmware 2.1.1.2 or later)
- 5312
- 5320/5320e
- 5324
- 5330/5330e
- 5340/5340e
- 5360

See the *MBG Remote IP Phone Configuration Guide* (available on the Mitel eDocs website at http://edocs.mitel.com) for details on setting up these phones through a connected MBG.

See the *Mitel Border Gateway Online Help* and the *Installation and Maintenance Guide* (both available on the Mitel eDocs webstie at http://edocs.mitel.com) for details about installing and configuring an MBG.

Both SIP devices and SIP trunking are supported through a connected MBG.

SysAd and Database Programming are not supported through a connected MBG.

System Licensing Requirements

Software licensing controls device capacity and feature availability on the MiVoice Office 250. Licensing is associated with a system by means of a serialized compact flash-type card or a USB key that inserts into a recessed USB slot located on the front of the unit chassis.

MiVoice Office 250 systems are delivered without software licensing. Licenses are generated and downloaded from the Applications Management Center (AMC). By default, unlicensed MiVoice Office 250 systems can be programmed with the capabilities of fully provisioned systems. However, if the applicable licenses are not provided and authenticated within four hours, the system automatically reboots itself.

If the system does not have a software license, Alarm 125 is sent to the LCD panel on the unit chassis and to the administrator phone. This alarm is generated every 5 minutes, and the system reboots every 4 hours until the license issue is resolved. For detailed information about system alarms, refer to the latest *Message Print Diagnostics Manual*.

If DB Programming connects to a system with a valid software license, you cannot restore a database that would invalidate the license. If you need to change the product type, you must first load a new software license through DB Programming. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Compact Flash Licensing Information

Starting with the v3.2 software release, the system now supports Base Server chassis licensing through a serialized compact flash-type card rather than through a USB key.

NOTE Whereas systems without a PS-1/vPS-1can now use either USB key licensing or compact flash-type licensing, the physical PS-1 chassis must still use USB key licensing.

All features currently licensable using a USB key can be made available for licensing using a compact flash-type card. The license file from the AMC can be associated with either medium, but not both concurrently.

Specifics

The following list provides some specific information about compact flash-type licensing:

- Requires system software v3.2 or later.
- Contains the same feature licensing information as was used on the USB key.
- Applies only to systems not equipped with a PS-1/vPS-1. Systems with a physical PS-1 will continue to use a USB key for licensing.
- Existing systems upgrading to v3.2 or later can continue to use a USB key for licensing. New systems will ship with a serialized compact flash-type card for licensing.
- When adding a PS-1/vPS-1 to a system already using the newer compact flash-type licensing, a USB key is once again required for licensing. The licensing information on the existing compact flash-type card will not be used.
- Licensing via a USB key can be replaced with licensing via a serialized compact flash-type card, but it is not necessary to do so. If an existing USB key fails, a replacement USB key can be used or a serialized compact flash-type card can be used.

- The licensing information is based on the media type. If using a USB key, a license file must be generated and match the USB key. If using a serialized compact flash-type card, a license file must be generated and match the compact flash-type card.
- How licenses are uploaded in DB Programming does not change regardless of using a USB key or a compact flash-type card.
- After using a compact flash-type card for licensing information, replacing the compact flash-type card requires re-licensing.

Upgrading

Although it is not necessary to change the licensing media when upgrading to v3.2 or later, if you choose to do so, you will need one of the serialized compact flash-type cards listed in the part numbers table on page 371. Note that there are different version memory cards for v3.2 and v4.0. This is because the core Linux file system on the card changed in v4.0. It is possible to upgrade an existing v3.2 card to v4.0, but it is not possible to downgrade an existing v4.0 card to v3.2.

NOTE Once the core Linux files have been downloaded and upgraded for v4.0 or later, there is no need to download them again for future v4.x or later system software upgrades.

Security

The serialized compact flash-type card or USB key uniquely identifies a system by storing a Mitel Master License. Only the specific software license that has been digitally signed with the Master License data is allowed to control the system. The card or key stores the essential components of the software licensing scheme.

Licensing Integrity

The security mechanism (whether it be a serialized compact flash-type card or a USB key) also ensures licensing integrity. When a system detects removal of its security mechanism, the system disables audio for subsequent calls. When the audio is disabled, the system posts applicable alarms to the LCD panel on the front of the unit chassis and to the LCD on the administrator phone. Alarm #122 indicates that the security mechanism is missing, while Alarm #125 identifies the number of minutes before the system initiates a reset.

When the mechanism is replaced and the correct serial number is detected by the system, the audio is enabled and the reset countdown is aborted.

If the system powers up without the security mechanism in place, the user must reload the software license through DB Programming.

USB Security Key for the PS-1/vPS-1

The system requires that a USB security key reside in the physical PS-1 unit. For vPS-1, the USB key must reside in the physical computer running the virtual machine. However, the Base Server does *not* need a USB security key. The administrator must upload the appropriate software license to the PS-1/vPS-1. For procedural details about uploading a license, refer to the *MiVoice Office 250 Features and Programming Guide*.

SIP Trunks License

Each SIP trunk requires a license. For 5000 v3.0 and earlier, SIP trunks used the IP Gateway Trunks license, part number 840.0234. For 5000 v3.1 and later, SIP trunks use the SIP Trunks license, part number 840.0833. This was used for connections to a SIP gateway. The newer SIP Trunks license replaces the older IP Gateway Trunk license. When you upgrade to v3.1 or later, an IP Gateway Trunk license converts to the newer SIP Trunks license. New installations of v3.1 or later using the SIP peer for trunk groups must purchase one SIP Trunks license for each SIP peer trunk. Each SIP trunk allows one concurrent call. The newer SIP Trunks license is used for connecting to a SIP gateway or for Voice Over Internet Protocol (VoIP) access.

Unified Voice Messaging

Each Unified Voice Messaging port is licensed, including the first four. Four UVM port licenses are included. Additional 4-port licenses must be purchased in order to use additional UVM ports.

Phone Model-Specific Licensing

All IP phones are licensed in accordance with three licensing categories. **Table 1** summarizes the categories and the phones requiring a license in each category.

Table 1: Phone License Categories

License Category	Phones Covered
Category A Previously licensed as Advanced IP Phones	• 8600, without display, in IP mode • 8622, 2-line display (and discontinued 8620), in IP mode • 8662, 6-line display, in IP mode • 8690 in IP mode Non-Mitel phone: OAISYS Net Phone, which emulates an 8662 in IP mode
Category B	8602 IP softphone
Category C	• 8600, without display, in SIP mode • 8622, 2-line display (and discontinued 8620), in SIP mode • 8662, 6-line display, in SIP mode • 8690 in SIP mode Phones that did not require an individual license in previous versions: • 8524, 8525, and 8526 wireless digital telephones • 8664, 8665, and 8668 wireless IP phones • 8660, 6-line display, IP mode only • IP PhonePlus • IP Single-Line Adapter (IP SLA) Other: Audio and Web Conferencing (AWC) port
Category D	52xx/53xx phones (5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, and 5360)
Category E	Digital telephones
(Sheet 1 of 2)	

Table 1: Phone License Categories (Continued)

License Category	Phones Covered
Category F	SIP phones
	(Sheet 2 of 2)

IP Gateway Trunk Licensing

The Inter-Tel 5000 platform requires “IP Gateway Trunk License,” part number 840.0234, for the use of the following two types of IP gateway trunks. One license is required for each trunk to be used.

- Media Gateway Control Protocol (MGCP)
- Session Initiation Protocol (SIP)

Software Feature Licensing

Software features, which customers can purchase to enhance their systems, are controlled by licensing through the Application Management Center (AMC). This section discusses the process used for licensing software features and the storage of multiple licenses on a system.

General

Because features are controlled through licensing, the system database cannot be restored with software features that have not been licensed by Mitel. If an attempt is made to Restore a database that has more software features than the license reflects, a warning message is displayed before the Restore command is executed. When this message displays, the Restore action must be cancelled. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Because the software license determines the features that can be enabled, changes cannot be made directly in DB Programming. The process to add software features involves purchasing the license, building the licence from the AMC, generating the license, downloading it to the field PC, and installing it in the system from the PC through DB Programming. For complete procedural instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

Number of Licenses

The system stores up to five different licenses, each of which is unique based on the system version. For example, only one v1.0 software license can be stored on the Processor Module. If a new license that matches the version number of an existing license is loaded, the new license automatically overwrites the existing license.

In addition, if five licenses are currently stored and a sixth license is saved, the oldest license, based on the date and time the license was created, is overwritten. If, however, the oldest license is currently in use, the next oldest license is overwritten. In addition, because each license is unique, only one license for each version number can be stored at any time.

IP Phone Licensing

This section identifies the types of IP phones that the Inter-Tel 5000 platform supports and the process used for activation of licensed phones.

Phone-Specific Licensing

The MiVoice Office 250 supports a three-tier licensing plan for IP phones. Category A licensing primarily allows use of Mitel Advanced IP phones. Category B licensing allows use of the Mitel 8602 IP softphone. Category C licensing primarily allows use of legacy 86xx phones. Category D licensing allows use of the Mitel 5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, and 5360 IP phones. For details, see Table 1, “Phone License Categories,” on **page 2-47**.

Types of IP Phones

The following list identifies the types of IP phones that operate on the Inter-Tel 5000 platforms. For specifications and illustrations of the phones, see “Phones” on page 101.

- IP mode only, including 5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, and 5360. Also includes 8602, 8660, 8664, 8665, 8668, and Multi-protocol (IP only), including 8600, 8620, 8622, 8662, and 8690
- IP Single-Line Adapter

Phone License Activation Process

Although you can program IP phones into the database without having an adequate number of licenses, the system allows only the number of phones to come online for which licenses exist. The system assigns licenses to phones on a first-come, first-served basis. However, once the available licenses have been assigned, the system prevents additional IP phones from coming online. As additional IP phone licenses are loaded into software, the system brings programmed but unactivated phones online. For detailed information about setting-up IP phones for service, see the **Installation** chapter, and refer to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*.

When IP phones fail to come online, the Message Print log identifies the excluded phone extensions. At the same time, Alarm #127—Unlicensed IP device failed to come online—appears on the unit chassis LCD panel and on the administrator phone display(s). For details, refer to the *Message Print Diagnostic Manual*, part number 550.8018.

Digital Telephone Licensing

The system can support up to four DEI units, and each unit supports up to 48 digital telephones.

Instead of using Digital Expansion Interface (DEI) licensing, digital telephones are now licensed the same way that IP phones are licensed in the system. Starting with v4.0, a new category E license has been added for digital telephones. When upgrading existing systems to v4.0 or later, the AMC will convert the old DEI licensing model to the category E licensing model. Each DEI license is converted to 48 category E licenses. Note that SLAs attached to digital circuits do not require a license.

Because the DEI units are identical, the distinction between them is determined by the cabling order and selection in DB Programming. The unit that is cabled directly to the Base Server is the first unit, and the unit that is cabled to the first DEI unit is the second unit, and so on.

System Alarms for Insufficient Licensing

Attempts to use IP phones or digital telephones programmed in DB Programming without having appropriate IP phone or digital telephone licenses will trigger Alarm 127, and the phone will not come online. Mitel recommends purchasing the appropriate phone license for each phone installed on the system. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

IP Resources

On a Inter-Tel 5000 platform, IP resources are required for the following communication devices and services:

- IP phones
- IP network connections
- Faxing between MiVoice Office 250 nodes through Mitel private networking over IP

IP resources are dynamically allocated where needed, which allows oversubscription. However, resources can be reserved if necessary. For details, see **Oversubscription**, below.

IP Resources Defined

An Internet Protocol (IP) resource is a quantity of processing power that digital signal processors (DSPs) on the system use for call processing functions. In terms of processing power consumed, the “cost” of a DSP function varies, depending on whether the DSP resides on the Processor Module (PM-1) or on the Processor Expansion Card (PEC-1) in the Base Server. IP resources are used for DSP processing of speech vocoder and Caller ID functions.

Expanded System Capacity

To achieve greater system capacity, a new installation uses G.711 as the default vocoder instead of G.729. Upgraded systems retain their previous configurations.

Mitel recommends that you re-evaluate the use of vocoders in call configurations after upgrading a customer’s system to v2.0 or later. To improve voice quality and expand capacity, you should use G.711 for internal IP phones and G.729 for external IP phones.

Oversubscription

Oversubscription allows you to program more phones than there are IP resources available on the MiVoice Office 250. This capability is useful for systems that support numerous IP phones and IP trunks and that utilize IP networking. The ability to oversubscribe results from the following interrelated factors in a MiVoice Office 250 v2.0 or later platform:

- IP resources are required only for call-related activities, which allows the unreserved resources to be pooled and shared for general use.
- IP resources are *not* reserved when an IP phone comes online, as was the case with earlier MiVoice Office 250 software versions. Dynamic allocation of IP resources is the only method available for allocation. You can no longer allocate IP resources statically in DB Programming. However, in oversubscribed systems, you can use the Resource Reservation tool to dedicate resources for specific IP devices. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.
- The Resource Reservation tool allows you to reserve, or allocate, IP resources for only the highest priority requirements. By default, IP resources are reserved for Emergency/911 access. However, in some configurations, additional resources may be required to accommodate IP gateways and/or IP trunks in order to reach an Emergency/911 operator. Otherwise, a customer may want to reserve a limited number of IP resources for additional high-priority functions such

as IP networking or UVM. In the more effective scenarios, however, the majority of IP resources would be kept available to be shared as needed. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Together, the preceding factors allow you to configure more IP phones than the number of IP resources available in a system. If all IP resources are in use, subsequent demands for IP resources are placed in a Camp On state until IP resources become available.

Depending on the configuration of reserved IP resources, a few scenarios may produce an unavailable IP resource condition. When system configuration results in an unavailable resource, a warning appears on display phones, and the user hears a beep every 5 seconds. To review the scenarios that may generate an unavailable IP resource error, refer to the *MiVoice Office 250 Features and Programming Guide*.

NOTE To effectively oversubscribe an IP-dominant system, the Resource Reservation feature must be used sparingly to avoid depleting the system pool of available IP resources. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.

IP Device Resource Manager

The IP Device Resource Manager maintains all IP resource requests in the system along with all allocated IP resources in the system. It provides useful diagnostic data to determine what types of calls are going on in the system and how long calls have been both camped-on or on an active call. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Node IP Connection Group

The Node IP Connection Group corresponds to an IP network connection between a remote node and the local node.

Networking

Two or more MiVoice Office 250 systems can be connected with T1/E1/PRI network spans to form a network. To the user, the network appears as one integrated system. With few exceptions, the user can perform all of the functions across the network that can be performed within a single system, or node. For networking specifications and details, see **Private Networking**.

In a Mitel IP private network environment, IP phones can be programmed to traverse, or bypass, the Network Address Translation (NAT) tables. A MiVoice Office 250 may be placed behind the firewall using a static NAT assignment.

For details about networking options, see **Network IP Topology**. For procedural instructions to implement near-end NAT traversal for IP phones, refer to the *MiVoice Office 250 Features and Programming Guide* and “Configuring Near-End NAT Traversal for IP Phones” on **page 1-431**.

Unified Voice Messaging

With applicable licensing, Unified Voice Messaging (UVM) provides up to 32 ports of voice messaging as follows:

- 24 UVM ports (maximum) on a standalone CS Controller (without a PS-1/vPS-1)
- 32 UVM ports (maximum) on a standalone HX Controller (without a PS-1/vPS-1)
- 32 UVM ports (maximum) on a CS or HX Controller equipped with a PS-1/vPS-1

For comprehensive details about UVM, refer to the *MiVoice Office 250 Features and Programming Guide*.

Optional External Messaging Systems

Beginning with the v3.0 release, the Voice Processing Unit (VPU) is not supported on the MiVoice Office 250.

For increased capacity and additional functionality, you can replace UVM with a NuPoint Unified Messaging (UM) system. NuPoint UM systems can be installed on any or all nodes in the network.

NuPoint UM is an optional, external system that unifies communication and messaging by providing access to voice mail, fax, and e-mail messages in the user's mailbox. For detailed information, refer to the NuPoint UM documentation located on the Mitel eDocs Web site (edocs.mitel.com).

NuPoint UM systems allow a caller to leave (on a local system) a message addressed to a mailbox located on a different system in the network. NuPoint UM has its own internal message queue, similar to a mailbox message queue, and stores messages until they can be delivered.

NOTE The optional Mitel Messaging IP Networking software license is required to use Voice Processing Nodes. Refer to the *MiVoice Office 250 Features and Programming Guide*.

For additional information on NuPoint UM, refer to the following documents:

MiVoice Office 250 Features and Programming Guide, part number 580.8001.

MiVoice Office 250 and NuPoint UM Integration Guide, part number 580.8008.

Hardware Summary

The **Specifications** chapter identifies preinstallation requirements and environmental parameters for optimal system performance and defines hardware and software specifications. The scope of the chapter includes specifications of the chassis, the compact flash-type memory card, expansion modules, phones, and interfaces for peripheral equipment.

NOTE **Compact flash** is a type of data storage device packaged in a matchbook-sized card (1.7 in. [43 mm] x 1.4 in. [36 mm] x .13 in. [3.3 mm]). CompactFlash® is a registered trademark of the SanDisk Corporation and is licensed through the CompactFlash Association. In this document, however, "compact flash" is used in the general sense to describe the type of data storage device that is common in the marketplace. "C Flash" is a shortened form of the term that appears on Mitel memory card packaging.

Features

This section summarizes MiVoice Office 250 software features—standard features that come with every system and software license features that are available at an additional cost or require additional licensing. For comprehensive descriptions and feature operating instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

Standard Features

The following standard features are included with each MiVoice Office 250:

- **Advanced Central Office (CO) Interfaces**: The advanced trunk features in this feature include Basic Rate Interface (BRI), Integrated Services Digital Network (ISDN), Primary Rate Interface (PRI) Private Network, and T1/E1/PRI. (The second port of a T1M-2 module requires additional licensing.)
- **Automatic Route Selection (ARS)**: ARS is a money-saving feature that allows the system to be programmed to select the least expensive route for placing outgoing calls. For ARS programming information, refer to the “Numbering Plans” chapter in the *MiVoice Office 250 Features and Programming Guide*.
- **Basic Call Processing Features**: The basic call processing features include standardized business features, such as Call Forwarding, Call Transfer, Call Waiting, Conferencing, Do-Not-Disturb (DND), Redial, Speed Dial, and so on.
- **Unified Voice Messaging (UVM)**: Unified Voice Messaging (UVM) provides basic voice messaging services and optional E-Mail Synchronization for system users. UVM runs on the Linux operating system and does not support Fax-On-Demand, or buffered SMDR.
- **Directories**: The Directory feature enables display phone users to “look up” intercom extension numbers and user names, System Speed Dial numbers and names, or system feature codes.
- **Multilingual Support**: Voice Processor prompts, phone displays, directories, DND messages, and reminder messages are available in American English, British English, Spanish, Canadian French, and Japanese. The database contains a programmable Language field for each trunk and phone to determine the default language. Also, a Change Language feature code can be entered by a phone user at any time to change the language.
- **Networking**: The MiVoice Office 250 can operate as a stand-alone system or as a node in a network of up to 63 MiVoice Office 250 nodes.
- **Record-A-Call**: This feature allows a phone user to record an ongoing call and place it in a voice mailbox.
- **System Forwarding**: System forwarding provides the ability to program the system so that calls ringing at a phone will follow a database-programmed “forwarding path” that routes the call based on the type of call and the status of the intended phone.
- **Uniform Call Distribution (UCD) Hunt Groups**: UCD hunt groups provide additional features to improve hunt group efficiency. When the UCD feature is enabled, additional system capabilities include announcement and overflow stations, hunt group priority ranking, and hunt group supervisors.

Software License Features

The following features require a software license.

- **Automatic Call Distribution (ACD) Hunt Groups**: Automatic Call Distribution can be programmed to distribute hunt group calls to equalize call time or call count among the available members. ACD can also provide call information records that can be processed by an external device connected to a system serial port.
- **Additional T1/E1/PRI Ports**: This license determines how many additional Dual T1/E1/PRI ports can be used on the system.
- **Agent Help**: With Agent Help, a phone user can request help from a supervisor while on a call. The supervisor can then choose to monitor or join the call, or to reject the request.
- **Analog Voice Mail Hunt Groups**: A hunt group can be designated as an analog voice mail hunt group to enable it to contain the multiple ports of an analog voice mail unit.
- **Unified Voice Messaging Ports**: This license determines how many Unified Voice Messaging ports can be used on the system.
- **Category A, B, C, D, E, and F Phones**: These licenses determine the number of multi-protocol and multimedia, multi-protocol touch screen phones that can operate on the system.
- **Desktop Interface**: Desktop Interface is an Open Architecture Interface (OAI) feature that allows devices such as PC phones, operator consoles, ACD supervisors, and integrated contact management tools to be developed for the MiVoice Office 250.
- **Dynamic Extension Express**: This license determines whether the Dynamic Extension Express software license is uploaded to the system.
- **File-Base MOH Sources**: This license determines how many File-Based MOH sources can be used on the system.
- **IP Gateway Trunks**: This license determines the number of IP gateway trunks that can be connected to the system. When upgrading, IP Gateway Trunk licenses get converted to SIP Trunks licenses.
- **IP Networking**: This license determines how many IP networking trunks can be used on the system.
- **Primary Rate Interface (PRI)**: PRI is an Integrated Services Digital Network (ISDN) version of a T1 circuit. A PRI circuit provides 23 bearer (B) channels and 1 data (D) channel (30B and 2D in Europe) and can transmit data at up to 1.544 Mbps (2.048 Mbps in Europe). To use PRI on the MiVoice Office 250, a T1M or T1M-2 module is required in addition to the license. The license for this feature controls the number of PRI modules operate in the system.
- **Remote Automatic Call Distribution (ACD) Hunt Groups**: This license enables members of ACD hunt groups to reside on different nodes. When using Agent IDs, each ID number must be treated as existing throughout the network, so the same Agent ID numbers must be established on all nodes on which hunt group members reside or log on. In order to use this feature, a software license is required for both this feature and the Automatic Call Distribution Hunt Groups for all nodes on which members reside or log on.

- **SIP Trunks**: The SIP Trunk license indicates how many SIP gateway trunk licenses are uploaded to the system. If you are upgrading your system, your current IP Gateway Trunk licenses will convert to the SIP Trunks license.
- **SIP Voice Mail Ports**: This license determines how many SIP peer voice mail ports can be used on the system.
- **System Open Architecture Interface (OAI) Events**: System OAI Events is a system-level communications protocol that allows call management tools to be developed for the system, such as the MiContact Center Office. Contact Mitel for more information regarding these products.
- **System OAI Third Party Call Control (Level 2 OAI)**: In addition to the call management tools allowed in System OAI Events, System OAI Third Party Call Control is a system-level communications TSAPI protocol that allows command-oriented tools to be developed for the system. Contact Mitel for more information regarding these products.
- **Voice Processor IP Messaging Networking**: This license allows voice processors to be connected in a network using the TCP/IP protocol.

System OAI (Open Architecture Interface)

System OAI is a system-level communications protocol that allows command-oriented tools to be developed for the system. System OAI is available in two levels: Events and Third Party Call Control. System OAI Events provides call status information to an external device. Its primary use is for Automatic Call Distribution products such as the MiContact Center Office. System OAI Third Party Call Control allows system command-oriented tools to be used on the system. Contact Mitel for more information regarding products and development materials for these features.

NOTE Level 2 System OAI applications can be used on networked systems, and some must be used in conjunction with a CT Gateway to process the network information correctly. For more information, refer to the *CT Gateway Installation Manual*, part no. 835.2161.

Chapter 4

Specifications

Chapter Overview

Correct installation and operation of the MiVoice Office 250 depends on a technician's ability to access accurate and complete information about the system. The following pages provide a composite of the system's environmental, physical, and electrical characteristics, as well as its computing and networking capabilities. Applicable industry, quality, and workmanship standards have been identified where appropriate.

Information in the chapter flows from general environmental parameters, installation hardware, and cabling requirements to specific details about component dimensions and characteristics, networking and power requirements, and power surge and power interruption protection measures.

Sections summarizing specifications for phones and peripheral equipment compose the final sections of the chapter.

Environmental Requirements

Table 2 shows the ranges of environmental conditions for operation and storage of MiVoice Office 250 equipment and phones. The Mitel 5000 can be placed on a stable, horizontal surface, wall mounted with brackets, or rack-mounted. For details, see page 140 and page 144.

NOTICE

Possible Malfunction Due to Overheating. Never exceed the maximum operating temperature listed in **Table 2**. When installing the chassis and phones, allow a sufficient margin for error in case of air conditioning failure, routine maintenance, plant shutdown, etc. As a general guideline, if the conditions are suitable for office personnel, they are also suitable for all equipment and phone operation. A properly controlled environment will help to extend the operating life of the equipment.

Table 2: Equipment Environmental Requirements

Requirements	In Operation	In Storage
Temperature		
Equipment Chassis	32 to 104 °F (0 to 40 °C)	+40 to 185 °F (–40 to 85 °C)
Station Instruments	32 to 113 °F (0 to 45 °C)	–40 to 185 °F (–40 to 85 °C)
Relative Humidity (Non-Condensing)	5–95%	5–95%
Altitude	Up to 10,000 ft (3048 m)	Up to 40,000 ft (12,192 m)

Programming Computer Requirements

With the introduction of System Administration & Diagnostics in v4.0, Database (DB) Programming is now installed as part of System Administration & Diagnostics along with supporting utilities. For complete information on this application, including hardware and software specifications for the computer running the application, refer to both the System Administration & Diagnostics Guide (part number 550.8125) and the help accessible from the application's user interface.

Software Version Compatibility

The MiVoice Office 250 requires the Linux operating system version included with the software release to ensure proper operation of version-related features. Failure to use the correct operating system version may result in system instability. Clicking the Update All button on the Administrative Web Portal page ensures that the correct operating system, call processing (CP) code, phone firmware, drivers, and associated software are installed.

System-Wide Specifications

Important system-wide specifications appear in the following subsections.

Loop Limits

Table 3 identifies the maximum distances from the Mitel 5000 (whether it be the original CS Controller platform or the newer HX Controller platform) to the various digital and single line phones that can be installed on the system.

Table 3: Loop Limits for Digital and Single Line Phones

Phones	Port Impedance	Loop Limit (24 AWG)
From Base Server chassis: • To a single line phone or playback device connected to the built-in port • To a single line phone or playback device driven by a SLM-4 port **From DEI chassis**: To a single line phone or playback device driven by a SLM-8	600 Ω	2000 ft (609 m) REN 3.0
From DDM-16 (HX only) or DEI chassis: • To digital telephones • To a digital telephone equipped with PCDPM and MDPM and connected to PKM/DSS unit(s) or SLA	62 Ω	1200 ft (366 m)
From an MDPM: To a single line phone	195 Ω	3800 ft (1158 m)
From an SLA: To single line phones or playback devices. Supports facility code OL13C.	800 Ω	15,560 ft (4742 m)

Power Supply

The 40 W power supply in the original CS Controller chassis and the 120 W power supply connected to the newer HX Controller chassis provide power to the Base Server chassis and modules. The chassis must be connected to an isolated, dedicated, 105–125 VAC, 15 A (minimum), 57–63 Hz, single-phase commercial power source through an uninterruptible power supply (UPS). The CS

Controller's 40 W power supply provides +5 VDC, 4 A output, and the HX Controller's 120 W power supply provides +24 VDC, 5 A output.

The 100 W power supply in the optional DEI chassis provides power to the DEI chassis and modules. For universal operation, each chassis *must* be connected to an isolated, dedicated, 90–240 VAC, 15 A (minimum), 50/60 Hz, single-phase commercial power source through a (UPS). The DEI power supply provides +24 VDC, 5.3 A output. For expanded applications, all required power transformers should be plugged into appropriate power sources. For details, refer to the equipment manufacturer's documentation.

NOTICES

Do ***not*** use an extension cord.

Do ***not*** plug any other equipment into the outlet where the UPS supporting the Base Server or DEI is plugged-in.

Do ***not*** defeat, or leave floating, the ground wire in the AC power cord.

In addition to the system power back-up, a rechargeable battery located on the system Processor Module backs up the database. In the event of a power failure or brownout condition, this battery supports the module for up to one day. The battery automatically recharges when power is restored to the module.

Surge/Spike Protection

To reduce the effects of AC voltage surges and spikes that may cause damage to the electronic components, Mitel recommends installing a surge/spike protector with the following specifications:

- Clamp voltage transients at 300 VAC within 5 nsec when exposed to waveforms as described in the ANSI/IEEE Standard C62.41-1980 (IEEE 587).
- Radio frequency interference (RFI) and electromagnetic interference (EMI) noise reduction by at least 20 dB at frequencies between 5 kHz and 30 MHz.

The Mitel DataNet CommSource® division sells a surge/spike protector that meets the recommended specifications—part number 900.0500.

Uninterruptible Power Supply (UPS) Connections

To avoid electrically induced performance problems, all associated MiVoice Office 250 equipment should be sourced from the same AC phase and ground reference. To assure that the electrical source is the same, plug MiVoice Office 250 equipment into the same UPS unit. A common electrical source is most critical with the Base Server and associated Digital Equipment Interface (DEI) units.

Because the optional Processing Server (PS-1/vPS-1) is IP networked, a common electrical source is not as critical. However, if wattage and outlets are available, the physical PS-1 should be maintained on the same UPS. The physical PS-1 does not require its own UPS unit.

Comply with the following electrical requirements when installing or upgrading MiVoice Office 250 equipment:

- Plug MiVoice Office 250 equipment into the same UPS unit, in particular the Base Server and associated DEI units. If wattage and outlets are available, the physical PS-1 should be maintained on the same UPS.

- The UPS unit must be plugged into an isolated, dedicated circuit providing required voltage, current, and grounding specifications.
- The power rating of the UPS unit must exceed the combined power ratings of all the equipment the UPS supports. **Table 4** provides an example of the wattage ratings in relation to the server equipment used. This information applies only to Mitel equipment.

Component systems having power supplies include the MiVoice Office 250 server, Digital Expansion Interface DEI(s), and the external voice processing systems. The systems plug into the battery back-up unit, which *must* be plugged-in to an isolated, dedicated circuit having the power specifications identified in “Power Supply” on **page 1-61**.

Table 4: Uninterruptable Power Supply Wattage Requirements

Equipment	UPS Wattage
CS Controller chassis only	60 W
HX Controller chassis only	120 W
With optional physical PS-1 chassis	add 200 W
With optional external voice mail	add 300 W
With each optional DEI	add 170 W

Mitel recommends installing a backup power supply unit having the following characteristics:

- Sine wave output
- Transfer time of less than 20 msec
- Low AC line transfer voltage, or cutoff circuit, of 100 VAC, minimum

The ampere-hour rating of the battery determines the system back-up time. Also, even if the power supply unit possesses the specifications listed in the preceding paragraph, it cannot be guaranteed that it will work properly with the Mitel system.

UPS Monitoring

For correct operation, the UPS Monitoring feature requires compatible hardware and software, as described in the following sections. For an expanded description of the UPS Monitoring feature and settings associated with UPS Monitoring, refer to the *MiVoice Office 250 Features and Programming Guide*. For installation instructions, see “Equipment to Support UPS Monitoring” on **page 1-148**.

Compatible UPS Units

Three commercial UPS units were tested against the Release 2.2 software and passed UPS Monitor-compliance testing for use with the MiVoice Office 250.

Monitoring capabilities of the following UPS units were found compatible at that time with the UPS Monitoring feature:

- APC Smart-UPS® series (Universal Serial Bus [USB] connection)
- MGE Pulsar Evolution series (USB connection)
- ONEAC ON® Series® in advanced interface mode (To connect, you must use the USB cable provided by the manufacturer.)

NOTE

Consider the following facts about the compatible UPS units:

- Only monitoring capabilities were tested.
- Only connection across USB was supported.
- Other models/series available from APC, MGE, and ONEAC were not supported.
- The best choice of a UPS for a particular system depends on environmental factors and a balance between rated load and desired runtime.

It is the responsibility of the organization and person(s) performing the installation and maintenance of MiVoice Office 250 systems to choose, test, and deploy a suitable UPS solution that both meets the customer requirements, and at the same time does not cause any problems with the normal operation of the MiVoice Office 250 or with the chosen UPS.

Main Distribution Frame (MDF) and Cabling

Connections between the trunks, digital telephones, external equipment, and system modules are made at the Main Distribution Frame (MDF). The MDF consists of a plywood backboard mounted on a wall and equipped with adequate interconnection hardware. Typically, the interconnection hardware includes industry-standard 66M1-50 terminal blocks, or 237A Krone blocks in Europe. Bridging clips are used on all terminal blocks to complete cross-connections. A patch panel and/or appropriate modular jacks may be used to connect other types of trunks to the chassis. **Table 5** summarizes and describes the materials used to construct an MDF.

Table 5: Main Distribution Frame Materials

Material	Description and Purpose
4 ft x 6 ft x 3/4-inch backboard, typical[a]	Mounted securely on a wall, the backboard provides the vertical surface for mounting terminal blocks, modular jacks, power strips for AC transformers, or other ancillary equipment.
In the United States, 66M1-50 Terminal Block In Europe, 237A Krone Block	May be used to terminate cables from the telephone company (CO) or Local Exchange to the customer's premises. For systems equipped with digital telephone capability, the blocks may be used to terminate cables from the Mitel 5000 to the MDF for distribution to end users.
Equipment rack	If using the rack-mount installation method, the rack should be of standardized dimensions for mounting telephone or data equipment. The rack should be set up so that the front of the chassis is at eye-level so the technician can read messages or commands appearing on the LCD panel. For further details, see the **Installation** chapter.
Patch panel	If using the rack-mount installation method, the patch panel should provide the appropriate number of ports for supporting the system. Each port shall provide an 8-pin/8-conductor (8P/8C) interface in a 110-type termination block suitable for 22–24AWG wire. For further details, see the **Installation** chapter.
Patch cables	If using the rack-mount installation method, patch cables terminated with 8P/8C plugs provide rapid, safe, and secure interconnection of trunk module circuits with telco facilities. For further details, see the **Installation** chapter.
6-pin to 8-pin step-up plug adapters	For connecting cables terminated with 6-pin plugs into 8-pin patch panel sockets.
6-wire modular jack assembly wired to RJ-11 specifications	For prewiring CO lines for Power Fail Bypass circuits.
6-wire modular jack assembly wired to RJ-14 specifications	May be used to terminate cables from the telephone company (CO) to the customer's premises. May also be used for prewiring CO trunks.

Table 5: Main Distribution Frame Materials (Continued)

Material	Description and Purpose
8-wire modular jack assembly wired to RJ-48C specifications, or to RJ-45S in Europe	For prewiring T1, T1/PRI, and E1/PRI trunks.
Four- or 6-wire modular plug	For terminating cable ends for insertion into jacks.
24 AWG twisted, unshielded cable[b]	Appropriate wiring for making network connections to telephone facilities and, where appropriate, to IP end users.
Bridging clips	To ensure complete electrical contact between cross-connected wires on a 66M1-50 or 237A Krone block.
Miscellaneous hardware, materials, and tools	Appropriate tools and fasteners to mount the backboard on a wall and to secure terminal blocks and other system-required equipment to the backboard.

a. System requirements will determine the size of the MDF backboard and the quantity of each item.
b. Provide sufficient Category 5 (CAT 5) or CAT 5e cabling for connecting the LAN server to the Mitel 5000 and for connecting IP phones to the LAN, as needed.

Cabling to Phones

IP phone cabling consists of Category 5e (CAT 5e) or CAT 5 cable connected between the IP phone and the local area network (LAN) at the user's phone location.

A 10 ft. (3 m.) 25-pair cable is available for connecting the HX Controller's DDM-16 to the main distribution frame (MDF). The cable, part number 813.1910, is terminated on one end with a special connector that attaches to the 48-pin connector on the DDM-16. The other end of the cable is terminated with a male Amphenol-type connector that attaches to the station block at the MDF. For usage instructions, see "Connecting Station Cables to the MDF" on **page 1-233**.

For the DEI DEM-16 station cabling information, see page 96.

Connections to the Public Network

In the United States, CO trunks are terminated on telephone company 66-type connecting blocks, as required by FCC regulations. Cross-connect cable connects the 66-type blocks to the CO terminal blocks on the MDF. The trunks are then connected to the Loop Start Modules (LSM) using cable terminated with 8-pin plugs wired to RJ-14 specifications. An alternate method is to interconnect the lines through a patch panel. In Europe, Local Exchange Trunks are terminated on 237A Krone blocks.

DID trunks are terminated on telephone company RJ-type modular jacks (or on 66-type blocks that are then cross-connected to modular jack assemblies), as required by FCC regulations. Two-pair, nonreversing, mod-to-mod line cord is used to connect each DID trunk to the appropriate jack on the corresponding SLA.

T1 spans and Primary Rate Interface (PRI) spans are terminated on RJ-48C modular jacks or on 66-type connecting blocks that are then cross-connected to modular jack assemblies, as required by FCC regulations. Four-pair, nonreversing, mod-to-mod line cord is used to connect each T1 and/or PRI span directly to the T1 jack on the corresponding T1M or T1M-2 module.

Mitel recommends installing gas discharge tubes with silicon avalanche suppressors on all non-T1/PRI trunks for lightning protection. For T1, T1/PRI, and E1/PRI spans, the Channel Service unit (CSU) itself provides the lightning protection. Also, in areas with frequent occurrences of

lightning, it is recommended that the cable between the telephone company termination and the gas discharge tubes be at least 75 ft (22.5 m) in length. If desired, the cable may be coiled.

Table 6:

Characteristic	Specification
Loss from CO to phone	0dB (@1 kHz, 0 ft)
Loss from CO to single line set	0dB (@1 kHz, 0 ft)
Loss from CO to CO	0dB (@1 kHz, 0 ft)
Ringer equivalence	0.0B REN
Ringing voltage	40 V_{RMS} minimum
Ring frequency	17–63 Hz
Loop current	18 mA, minimum
Protection, tip-to-ring	400 V transient

Capacities

For general reference purposes, the topics and tables in this section summarize the MiVoice Office 250 equipment specifications and capacities.

NOTICE

The capacity values published in this document have been validated under controlled laboratory conditions and supported by real-time field trials in a variety of business operations. However, the actual capacities that a MiVoice Office 250 can support will depend on the mixture of trunks and phones installed, among other network and environmental factors.

While Mitel converged technology platforms provide advanced communications capability, the nature of that technology imposes certain physical constraints. These physical constraints include such things as the number of simultaneous voice channels and signal processing resources available at any time. The maximum capacities listed indicate tested capability rather than claims that all the values have been realized at the same time.

System Capacities

Table 7 summarizes capacities for systems running v6.0 or later software.

NOTE Starting with v4.0, the lack of a PEC-1 and/or a PS-1/vPS-1 no longer reduces certain maximum feature capacities available on the system. For guidelines on when they might be needed, see the MiVoice Office 250 Engineering Guidelines.

Table 7: System Capacities for Systems Running v6.0 or later

System Component	CS Controller	HX Controller
Absorbed digits		
• Patterns per system	50	50
• Digits per pattern	48	48
Account Codes		
• Forced	512	512
• Standard	512	512
• Digits per code	3–12	3-12
Analog Ports (built into the main chassis)		
• Loop Start	2	4
• Single Line	2	4
Automatic Route Selection (ARS)		
• Route groups	100	100
• Facility groups per route group	100	100
• Trunk groups per facility group	208	208
• Dial Rules (16 digits each)		
• Total	32	32
• Programmable	26	26
Basic Rate Module (BRM-2) per system [UK only]	3	4
Busy Hour Call Completion (BHCC)	8000	8000
Callback Queue		
• Requests per system	250	250
• Requests per phone	1	1

(Sheet 1 of 4)

Table 7: System Capacities for Systems Running v6.0 or later(Continued)

System Component	CS Controller	HX Controller
Camp On		
• Camp On by a phone	1	1
• Camp On requests per system	250	250
• Phones camped on to a phone	250	250
• Phones camped on to a trunk group	250	250
Class of Service (COS) designations per phone	16	16
Conferencing Resources -- Basic Ad Hoc		
• Total per system	20	20
• Parties per conference	4	4
• Simultaneous four-party conferences	5	5
Conferencing Resources -- Advanced Ad Hoc[a]		
• Total per system	40	40
• Parties per conference	20	20
• Simultaneous twenty-party conferences[b]	2	2
Conferencing Resources -- Meet-Me		
• Total per system	40	40
• Parties per conference	20	20
• Simultaneous twenty-party conferences[c]	2	2
Digital Desktop Module (DDM-16)	N/A	3
Digital Endpoint Module (DEM-16) per DEI		
• Per DEI	3	3
• Per system	12	12
Digital Endpoint Module Ports per DEM-16	16	16
Digital Endpoint Module Resources		
• DTMF receivers	4	4
• Speakerphone	16	16
Digital Expansion Interface (DEI) Units	4	4
Digital Station Select/Busy Lamp Field (DSS/BLF) Unit or Programmable Key Module (PKM)		
• Per digital telephone (depends on phone type)	2-4	2-4
• Per system, or node	20	20
Do-Not-Disturb (DND)		
• Message options per phone	20	20
• Messages per system	250	250
(Sheet 2 of 4)		

Table 7: System Capacities for Systems Running v6.0 or later(Continued)

System Component	CS Controller	HX Controller
Dual Tone Multi-Frequency (DTMF) Resources		
• Receivers	6	6
• Transmitters	4	4
Extension Lists		
• Per system	100	100
• Phone or trunk numbers/list	250	250
• Phones/list used for Hunt Groups	60	60
• Phones/list used for Paging	60	60
Fax over IP (FoIP)—T.38 (Maximum simultaneous faxes between MiVoice Office 250 systems only)	6	6
Forwarding Paths		
• System total	200	200
• Paths per phone	3	3
• Points per path	4	4
Hunt Groups per system	75	75
	300 (In case of PS1 System)	300(In case of PS1 System)
Keymap Group Assignments		
• Phone	25	25
• PKM or DSS/BLF (in DB Programming only)	20	20
• User-programmable buttons per phone	35	35
Loop Start Module		
• Dual (LSM-2)—No longer in production.	3	4
• Quad (LSM-4)	3	4
Music On Hold port per system	1	1
Paging		
• Port per system	1	1
• Zones per system	10	10
• Phones in a paging zone	60	60
Phones		
• IP	250	250
• Digital	192	240
• Single-Line	102	112
Reminder Messages		
• Programmable messages	20	20
• Message requests per system	250	250

(Sheet 3 of 4)

Table 7: System Capacities for Systems Running v6.0 or later(Continued)

System Component	CS Controller	HX Controller
Single-Line Adapter		
• Per Digital Endpoint Module	16	16
• Per Digital Expansion Interface	48	48
• Per system, or node*	48–96	48-96
• *With 2 single line phones per SLA	48	48
• *With 1 single line phone per SLA	96	48-96
Single Line Module, 4-port (SLM-4) [Only in Mitel 5000]	1	4
Single Line Module, 8-port (SLM-8)		
• Per Digital Expansion Interface	3	3
• Per system, or node	12	12
Socket connections per system		
• Attendant Consoles	23	23
• Desktop Open Architecture Interface (OAI)	30	30
• Level 2 OAI	3	3
• Message Print	3	3
• Station Message Detail Recording (SMDR)	3	3
• System Monitor	3	3
Station-to-Station messages per system	250	250
System Speed Dials [d]	1000	1000
T1/E1/PRI Module		
• Single (T1M) — No longer in production	3	4
• Dual (T1M-2)	3	4
Toll Restriction User Groups		
• Groups per system	8	8
• Area Code tables per User Group	1	1
• Area Codes per table	800	800
• External Area Codes per User Group	6	6
• Office codes per table	800	800
• Office code tables per external Area Code	2	2
Trunking		
• Trunk Groups	208	208
• Trunks per Trunk Group	208	208
• IP gateway trunks	32	32
Unified Voice Messaging (was Basic Voice Mail)		
• Ports per system	24 or 32[e]	32[f]
• Mailboxes enabled for synchronization — without a PS-1/vPS-1	100[g]	100
• Mailboxes enabled for synchronization — with a PS-1/vPS-1	250	250
Vocoder Voice Channels		
• G.711, maximum	250	250
• G.729, maximum	120	120
• T.38 (fax over IP) maximum	6	6
Voice Mail Ports (Voice channels)		
• Using Unified Voice Messaging (Requires additional licensing after first 4)	32	32
• Using NuPoint UM	32	32
(Sheet 4 of 4)		

a. If the Advanced option is enabled for Ad Hoc Conferencing and the system is licensed for Meet-Me Conferencing, the two conferencing applications share conferencing resources. See MiVoice Office 250 Enginerring Guidelines for more information.

b. Although the listed capacity is based on the twenty-party conference maximum (two twenty-party conferences), the system supports any combination of conferences up to twenty parties and 40 ports. For example, you could have four ten-party conferences or five eight-party conferences.

c. Although the listed capacity is based on the eight-party conference maximum (two eight-party conferences and one four-party conference), the system supports any combination of conferences up to eight parties and 20 ports. For example, you could have four five-party conferences or five four-party conferences.

d. Only 500 speed dials at a time can be programmed manually. 1000 Speed Dials can be imported at one time.

e. The CS Controller platform must be equipped with a PS-1/vP-1 to support up to 32 UVM ports. As a standalone system, up to 24 UVM ports are supported.

f. The HX Controller can support up to 32 UVM ports in either a standalone or PS-1/vPS-1 configuration.

g. PS-1/vPS-1 equipped platforms are not supported in Australia.

Licensing Capacities

Table 8 summarizes maximum licensing capacities for the system. Digital telephones require additional licensing.

Table 8: Maximum Licensing Capacities

Type of Device	Capacity
IP Phones Configured/supported	250
IP Trunks Configured/supported concurrently without IP Networking	32
IP Networking	
• With initial system	6
• With IP Networking license	250
Unified Voice Messaging (UVM) (4-port increments)	32[a, b]
Primary Rate Interface (PRI)	6 (CS), 8 (HX)
Second T1/E1/PRI Port	3 (CS), 4 (HX)
SIP Phones	250
SIP Trunks	100

a. The CS Controller platform must be equipped with a PS-1/vPS-1 to support up to 32 UVM ports. As a standalone system, up to 24 UVM ports are supported.

b. The HX Controller can support up to 32 UVM ports in either a standalone or PS-1/vPS-1 configuration.

Voice Message Storage Capacity

NOTICE

As of the v4.0 release, the system requires a minimum one GB compact flash-type memory card. If this system is currently using a smaller compact flash-type memory card, upgrade the card to one GB before upgrading the system to v4.0 or later.

Unified Voice Messaging (UVM) storage capacity has changed with the introduction of the G.726-32 option as an additional voice compression selection.

G.726-32 uses 32 kbps while the default G.729 compression uses 8 kbps. The default UVM compression is set to G.729 to maximize voice mail storage capacity, and the previously published voice mail capacities are based on G.729 compression. Using G.726-32 consumes four times as

much storage resulting in 25% of the storage capacity, but it also produces better audio quality for voice messages in some scenarios.

Systems equipped with a physical PS-1 do not offer a UVM vocoder selection because messages are stored on the PS-1 hard disk. Since storage is plentiful, no compression is used (G.711). There are some scenarios in which G.726-32 may be appropriate for UVM compression:

- In some environments where G.729 is used for IP phones, multiple compression/decompression steps may result in degraded voice quality of the stored messages. Using G.726-32 for voice mail storage may improve the quality of recorded voice.
- The VPIM feature, as specified in the VPIM standard, natively uses G.726-32 compression for the interchange between voice mail systems. By using G.726-32, the system does not need to transcode from G.729 to G.726-32. The result is more efficient processing.

If the resulting voice storage capacity is unacceptable, customers should upgrade to a larger compact flash-type card. Capacities are shown in **Table 9**.

Table 9: Estimated UVM Storage Capacity

Compact Flash-Type Card Size	Voice Mail Storage (non PS-1/vPS-1) using G.729 (default)	Voice Mail Storage (non PS-1/vPS-1) using G.726	Voice Mail Storage (with PS-1/vPS-1) using G.711
1 GB	6000 minutes (100 hours)	1,500 minutes (25 hours)	30,000 minutes (500 hours)
2 GB	18,000 minutes (300 hours)	4,500 minutes (75 hours)	30,000 minutes (500 hours)

NOTICE

Besides vocoder selection, items like freezes, log files, file-based MOH files, additional language prompts, etc. can reduce available UVM storage capacity. As a result, many installations may require a larger compact flash-type memory card in order to provide acceptable UVM storage capacity.

See the MiVoice Office 250 Enginerring Guidelines for more information on selecting the appropriate card size.

HX Controller

The newer MiVoice Office 250 architecture, referred to as the HX Controller, is designed to offer digital telephone support right out of the box, yet still provide adequate capacity for IP phones, trunks, and single-line devices. The hardware is detailed throughout this section. For installation instructions, see “Installing the HX Controller Base Server Chassis“ on **page 139**.

Chassis Description

The HX Controller is the newer Base Server chassis that houses the main processing module and the various station/trunk expansion modules. The 2U height chassis can be placed on a desk/table/shelf, mounted on a wall, or mounted in a standard rack.

The physical dimensions of the HX Controller chassis are listed in **Table 10** below.

Table 10: Dimensions of HX Controller Chassis

Dimension	Size
Height	3.5 in. (8.9 cm) — 2U
Width	15.5 in. (39.4 cm)
Depth	8.75 in. (22.2 cm)
Weight (approx. loaded)	9.2 lbs. (4.2 kg)

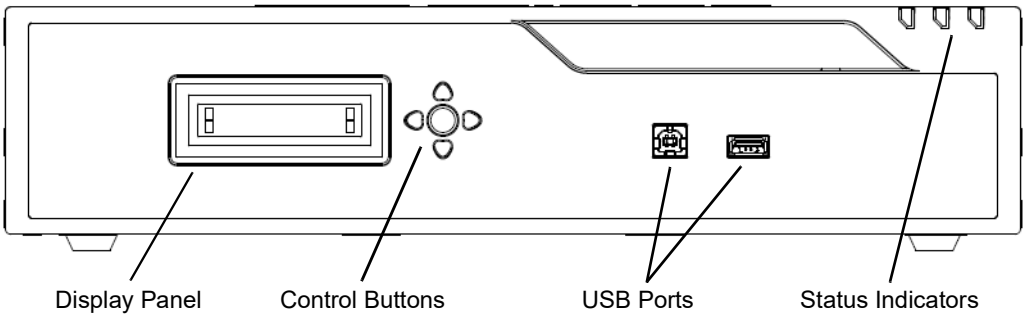
Front Panel

The front of the HX Controller chassis (see **Figure 1** below) provides the following:

- **Display Panel:** A two-row, 14-character per row liquid crystal display (LCD) panel to show system status and command messages.
- **Control Buttons:** A five-button, command-and-control interface for interacting with the system via the display. The control buttons allow you to scroll through menu items to set system parameters during system installation and to perform other system routines.
- **USB Ports:** One 2.0 type A connector (for database backup via a portable flash drive) and one 2.0 type B connector (for system online monitoring via a PC).
- **Status Indicators:** Three light-emitting diodes (LEDs) for indicating system status (as viewed from the front, left-to-right):
  - System Alarm (lit red if a system fault or failure is detected)
  - File System/Storage Access (lit green when accessing flash memory)
  - Power On (lit green when the chassis is receiving power; lit red if the chassis backplane is not functional)

The following illustration shows the front of the HX Controller chassis.

Figure 1: HX Controller Front Panel



Rear Panel

The rear of the HX Controller chassis (see **Figure 2** below) provides the following:

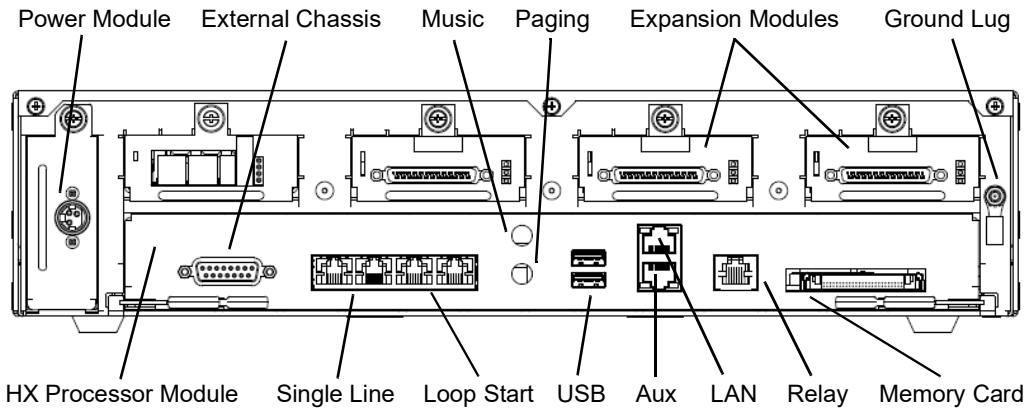
- **Power Module Bay:** Houses an internal power distribution module with a 3-pin DIN connector for connecting an external 120W, +24VDC power supply unit.
- **External Chassis Connector:** A DB-15 (15-pin D-subminiature socket) connector for attaching a DEI chassis.
- **Music Jack:** A 3.5 mm, two-conductor (stereo), mini phone jack for connecting an external music source that can be heard by callers when they are placed on hold or camped on, or used to provide background music to phone users. Due to hardware changes, this jack requires a slightly louder input than the jack on the CS Controller.
- **Paging Jack:** A 3.5 mm, two-conductor (mono), mini phone jack with an output impedance of 600 ohms for connecting external paging speaker equipment for paging to large areas like warehouses or loading docks.
- **Expansion Module Bays:** Four smaller bays for adding the 16-port Digital Desktop Module (DDM-16) or any of the existing station/trunk modules supported in the original CS Controller chassis.
- **Ground Lug:** For properly grounding the chassis.
- **HX Processor Module Bay:** A single large bay to house the HX Processor Module.
- **Single Line (FXS) Jacks:** Two six-pin RJ-14 jacks that allow up to four single line devices to be connected directly to the system without the need for single line expansion modules.
- **Loop Start (FXO) Jacks:** Two six-pin RJ-14 jacks that allow up to four loop start trunks to be connected directly to the system without the need for loop start expansion modules.
- **USB Ports:** Two 2.0 type A connectors for Uninterruptable Power Supply (UPS) monitoring or for connecting a USB hub to provide additional ports.
- **Auxiliary Port:** Reserved for possible future use.
- **Gigabit Ethernet LAN Port:** An auto-sensing (10/100/1000 Mbps) eight-pin RJ-45 jack for connecting the system to the customer's LAN or to a programming PC. It is configured to auto-negotiate both speed and duplex mode. The LED on the left side of the jack is lit for a

physical network link, and it flashes if the system detects activity on the link. The LED on the right is lit if the link speed is 1000 Mbps, and it is unlit if the link speed is 10/100 Mbps.

- **Relay Jack:** A six-pin modular jack for optional door relay control.
- **Memory Card Slot:** A single slot to house the Mitel serialized compact flash-type memory card. Next to the slot is a recessed extractor button for ejecting the card. (Use only Mitel-supplied memory cards; see page 371 for part numbers.)

The following illustration shows the back of the HX Controller chassis equipped with an HX Processor Module, a T1 expansion module, and three digital expansion modules.

Figure 2: HX Controller Rear Panel



Processor/Expansion Modules

In addition to the numerous older expansion modules that continue to work with the HX Controller platform, two additional modules were developed, as described below.

NOTE These two modules function only in the HX Controller chassis. Do ***not*** attempt to install them in the CS Controller chassis, as damage may occur.

HX Processor Module

As the name implies, the HX Processor Module contains the main controlling microprocessor and provides the core call and voice processing engine for the system. Key features include:

- 32-bit architecture
- Linux® operating system
- Built-in analog, digital, and VoIP (G.711/G.729) capability
- Built-in Unified Voice Messaging (UVM) capability (up to 32 ports in either a standalone or PS-1/vPS-1 configuration)
- T.38 fax over IP capability
- Power failure bypass function (via an internal cutover relay) to provide emergency calling capability in the event of a power failure

- Optional door relay control
- Real-time clock circuitry for accurate date and time function
- Digital signal processor (DSP) technology for conferencing resources, Dual Tone Multi-Frequency (DTMF) detection/transmission, tone generation, and modem operation
- Optional expansion for IP-heavy installations (or additional voice processing resources) via the same Processor Expansion Card (PEC-1) used in the original CS Controller chassis

16-Port Digital Desktop Module (DDM-16)

Up to 16 digital devices can be connected to the DDM-16, and there can be up to three DDM-16s installed in any of the four expansion bays. If more than 48 digital devices are required, up to four optional DEI units can be attached to the system. The same digital telephones and SLAs supported on the DEI's DEM-16 are supported on the DDM-16. Each DDM-16 includes:

- A single 48-pin connector and a special DDM cable for connection to a station block at the Main Distribution Frame
- Status LEDs (online, busy, and remove)
- Module release lever for safely removing the module from the bay
- DSP resources (4 DTMF receivers, 5 tone generators, and 16 speakerphones for devices connected to the module)

Note that the newer DDM-16s are not subject to any of the voice channel trade-offs that the original DEM-16s are subject to when used in two-DEI or four-DEI mode.

Door Relay Control

The HX Controller chassis includes a relay jack that can be used to unlock a door (i.e., "buzz" it open) by dialing a feature code. For details on hooking up a door relay control mechanism to the relay jack on the chassis, see "Installing the Door Relay Control" on **page 142**. To support this feature, there is a phone-related flag in DB Programming called Activate Door Relay (refer to the *MiVoice Office 250 Features and Programming Guide*, part number 580.8006, for details).

Support for Existing Expansion Modules

The same expansion modules available on the original CS Controller platform can also be used on the newer HX Controller platform, including:

- T1/E1/PRI Module (T1M) — discontinued and longer available for purchase
- Dual (2-Port) T1/E1/PRI Module (T1M-2)
- 2-Port Loop Start Module (LSM-2) — discontinued and longer available for purchase
- 4-Port Loop Start Module (LSM-4)
- Basic Rate Module (BRM-2) - UK only
- 4-Port Single Line Module (SLM-4) — no longer limited to one per chassis
- 8-Port Single-Line Module (SLM-8) — installed in the DEI chassis (see next section)
- 16-Port Digital Expansion Module (DEM-16) — installed in the DEI chassis (see next section)

The above expansion modules (except for the SLM-8 and DEM-16) can be installed in any combination in any of the four available expansion bays.

For detailed information on these existing expansion modules, see the appropriate information later in this chapter.

Support for Existing Chassis Expansion

The same chassis expansion options available on the original CS Controller platform are also available on the newer HX Controller platform, including:

- Digital Expansion Interface (DEI) chassis for additional digital or single-line device capacity via:
  - 16-Port Digital Expansion Module (DEM-16)
  - 8-Port Single-Line Module (SLM-8)

> **NOTE** If the HX Controller chassis and DEI chassis are rack mounted, connect them together using the newer Extended DEI Cable (50006488) instead of the existing standard-size DEI Cable (part number 813.1847). If the extended cable is not available, two standard DEI cables can be chained together to achieve sufficient length for rack mounting.

- Processing Server (PS-1/vPS-1) for additional processing power needed for many larger installations

For detailed information on these existing chassis expansion options, see the appropriate information later in this chapter.

Support for Existing Optional Equipment

The same optional equipment choices available on the original CS Controller platform are also available on the newer HX Controller platform, including:

- External Voice Processing (NuPoint Unified Messaging)
- Music Source
- Paging Equipment
- External Modem
- SIP Trunk Gateways
- CT Gateway

For detailed information on these existing optional equipment options, see the appropriate information later in this chapter.

CS Controller

The CS Controller chassis houses the system Processor Module (PM-1), optional Processor Expansion Card (PEC-1), the motherboard, and power supply.

Note: The CS controller is not supported from version 6.2.

Chassis Description

Table 11 summarizes the physical dimensions of the CS Controller chassis.

Table 11: Dimensions of Mitel 5000

Dimension	Measurement (Metric)
Length	14.50 in. (36.83 cm)
Width	17.00 in. (43.18 cm)
Height	01.75 in. (4.45 cm) (1 U)
Weight	09.00 lb (4.08 kg)

The front of the chassis presents a two-row liquid crystal display (LCD) panel and a five-button command-and-control interface for interacting with the system. The front of the chassis also provides two Universal Serial Bus (USB) ports, a USB socket for a software license security key, a bay that holds a compact flash-type memory card, and a recessed ejection button for extracting the card.

In addition to the PM-1 bay, the back of the chassis offers three bays that accept slide-in trunk modules and the following six connection interfaces:

- The system power cord socket
- A 1/8th-inch jack for attaching an audio source to supply system Music-On-Hold and/or Background Music
- An RCA-type phono jack for attaching an external paging unit
- A 6-pin socket for connecting two loop start trunks
- A 6-pin socket for connecting two single line phones.

  Together, the loop start and single line circuits also provide the system's power fail bypass feature.
- A DB-15 connector for attaching peripheral equipment
- The CS Controller chassis supports the following trunk circuit modules:
  - Loop Start Module, 2-port (LSM-2)—Supported but no longer in production.
  - Loop Start Module, 4-port (LSM-4)
  - T1/E1/PRI Module, 1- port (T1M)
  - Dual T1/E1/PRI Module, 2- port (T1M-2)
  - Basic Rate Interface Module, 2-port (BRM-2) [European market only]

Motherboard Elements

Circuitry and components on the CS Controller motherboard provide elements that are essential to the performance of the MiVoice Office 250, including:

- **System/real-time clock**: Provides accurate date and time for the system.
- **Built-in Network Clock Circuitry (NCC)**: The network clock circuitry provides the time division multiplex (TDM) clock for the entire system. This circuitry is necessary specifically when digital trunks (e.g., T1, T1/PRI, E1/PRI, BRI) are connected to the system. This circuitry allows the system to synchronize the system TDM clock to the outside world (telco) and provide a smooth, error-free clock transition when switching clock reference from one digital trunk to another.
- **Conferencing resources**: Supports up to 20 circuits in multiple voice conferences. The system allows up to five simultaneous four-party conferences or up to six simultaneous three-party conferences at a time. However, conferencing is *not* amplified.
- **DTMF receivers**: Provides six resources that convert DTMF signals to digital codes for use by the system. Used by the system when single line phones are dialing, when features such as DISA and analog voice mail are being used, and to receive Dialed Number Identification Service (DNIS) and/or Automatic Number Identification (ANI) digits for incoming T1 Ear-and-Mouth/2-way (E&M) calls. DTMF receivers are in use only while the dialed number is being processed, not for the duration of the call. If all DTMF receivers are busy, the user may try again. DISA users are redirected to the primary attendant.
- **DTMF transmitters**: Provides four resources for generating DTMF tones whenever needed by the system. For example, when a user manually dials a number while on a call or when a phone user uses one of the system auto-dial features (ARS, Call Forward to the public network, Station or System Speed Dial, etc.).
- **Tone generating circuit**: Produces all of the system tones (busy, reorder, Do-Not-Disturb, etc.) heard by phones, DISA, and automated attendant users.
- **Internal Modem**: Provides a dial-in modem that is accessible through Point-to-Point Protocol (PPP) and capable of transferring data at speeds up to 33,600 bps. This modem is always equipped and functions effectively as a system serial port over IP that the user must “call.” Because the connection is established with a call, each modem application in a system is assigned an extension. The default extension is 1500. For programming information regarding the modem, refer to the *MiVoice Office 250 Features and Programming Guide*.

Processor Module with Processor Expansion Card

The CS Controller Processor Module (PM-1) contains the main controlling microprocessor and associated control logic and memory circuits. The PM-1 is under the control of the IPR Application, which is stored in flash memory and activated when the system is powered up. An optional Processor Expansion Card (PEC-1) can be attached to the PM-1 to provide additional IP resources and increase the system’s processing capability. See the MiVoice Office 250 Enginerring Guidelines for details on when a PEC-1 might be needed.

A PM-1 provides the following system resources:

- **Database back-up battery**: (3.6V, lithium ion, coin cell rechargeable battery) Supports the database for at least 1 day of continuous system down time. On the PM-1 module, the LED labeled BAT ALM indicates the charge status of the battery. When the BAT ALM LED is not lit,

the battery is adequately charged. When the LED lights green, the battery is recharging. For instructions on replacing this battery, see “Replacing the Processor Module Battery” on page 250.

- **LED indicators**: Shows main processor operation and module removal indicators.
- **IP Port (100Base-T)**: An auto-sensing (10/100 Mbps) eight-pin RJ-45 jack provides IP connection directly to the system with CAT5 or CAT5e cable. In addition, the IP Port can be used to load Call Processing software or software licenses, or to program the database.

  This port is configured to auto-negotiate both speed and duplex mode. As such, the network switch port connected to this jack must also be configured to auto-negotiate speed and duplex mode.

  The LED on the right side of the RJ-45 jack is a link/activity LED. This LED is lit if the system detects a physical network link, and it flashes if the system detects activity on the link. The LED on the left side of the RJ-45 jack indicates the speed of the connection. This LED is lit if the link speed is 100 Mbps, and it is unlit if the link speed is 10 Mbps.

  NOTICE

  Security is the customer’s responsibility. Mitel is not responsible for network problems due to security violations involving the IP address exposure of the PM-1. This includes, but is not limited to, interrupted network service. Mitel recommends that the customer install the MiVoice Office 250 in a private IP network behind a firewall or Network Address Translation (NAT) unit.

- **Software Security and Feature Program**: Supports a software license that enables system functionality. For details about the software security and feature program, refer to the *MiVoice Office 250 Features and Programming Guide*.
- **Caller ID receivers**: Provides four resources for receiving Caller ID.

The PM-1 supports Call Processing across networks. Supports the United States and European versions of MiVoice Office 250 software.

Interfaces on Front of the CS Controller

The front of the CS Controller Base Server chassis is equipped with a two-line LCD panel, an LCD control consisting of four triangular buttons surrounding a circular button, two USB jacks, a Security Key socket, a recessed compact flash ejection button, and a compact flash-type card bay.

LCD Panel: The LCD panel displays system status and command messages on two lines of 14 characters each. provides two basic functions. It displays system status information that the technician uses to determine the operating condition of the system. As driven by five LCD control buttons, the panel displays menus and options for programming, changing, or resetting the system.

LCD Control Buttons: The control buttons allow the technician to scroll through menu items to set system parameters during system installation and to perform other system routines. The up and down arrow-shaped buttons control the movement of the menus and menu items displayed on the LCD panel. The up–down arrows also allow the technician to advance or retreat through letters or numbers. The right and left arrow buttons control the movement of the cursor from one character or command to another. Pressing the center, circular button causes the system to select and accept the chosen letter, number, or command item.

Manual System Reset: Reset is a software-controlled system reconfiguration, and the technician can manually reset the system using the LCD panel and control buttons. By selecting the **REBOOT** command from the SYSTEM MENU on the LCD panel and then pressing the center button, the technician causes the system to:

- Preserve the battery-backed database information, all calls in progress, outside calls being dialed, pages in progress, camped-on calls, Callback Queue requests, system and user programming, messages between phones, calls on system and individual hold, DND messages, and reminder messages.
- Restore DTMF receivers, modem access, and speech channels, and reset the phone clocks to match the system clock. Update button lamp status on PKMs and DSS units.
- Disconnect calls ringing in and calls in progress.

USB-B Serial Port: The USB (Universal Serial Bus)-B port connects to a personal computer for local accessing the online Monitoring (OLM) feature.

Security Key: This USB device contains a record of the system's licenses, allowing only authorized features to be implemented. With v3.2 or later software, the Base Server chassis can use a serialized compact flash-type card instead of a USB security key. See **page 45** for details.

Memory Card Bay and Extractor Button: This bay accepts a compact flash-type card distributed only by Mitel. The card stores system applications, the system database, and voice mail prompts and messages, including Record-A-Call recordings. For details, refer to the backup information in the *MiVoice Office 250 Features and Programming Guide*. Next to the bay is a recessed extractor button that ejects the card when pushed in. The button is recessed to reduce the risk of the card being unintentionally removed from the system.

Interfaces on Back of the CS Controller

The back of the CS Controller Mitel 5000 chassis consists of the connection interfaces for expansion units, the local area network (LAN), external paging, Music On Hold, the PM-1 bay, and bays for expansion modules.

Music-On-Hold Port: Connects an external music source (such as a radio, tape player, etc.) that can be heard by callers when they are placed on hold or camped on. It may also be used for phone background music. The port is an 1/8-inch (3.5 mm), two-conductor (mono), mini-phone jack. For installation instructions, see **page 329**. In Database (DB) Programming, music can be enabled or disabled, and each trunk group (as well as transfer audio) can be assigned music, silence, or tick-tones. Refer to the *MiVoice Office 250 Features and Programming Guide* for details. This port supports A-law (European) or mu-law (USA) companding at the system level, which is flagged in Database (DB) Programming (System\Flags\Companding Type).

If desired, one external music source may be connected to the system to supply background music and Music-On-Hold (MOH). The MOH jack is located on the rear of the system chassis.

MOH input circuitry is high-impedance (8.7 Kohm), which allows connection to any audio device with an output impedance of 4–8.7 ohms. MOH inputs are rated at 1 V_{RMS}. If input power exceeds this rating, audio may be distorted. Because of this, Mitel recommends limiting the input power to 1 W.

Six-pin jacks: Together the jacks provide the system's Power Fail Bypass feature through an internal CO cutover relay. One of the jacks supplies two loop start trunk circuit interfaces. The other jack supplies circuit interfaces for two single line devices.

External Paging Port: An RCA-type phono jack with an output impedance of 600 ohms. Connects external paging speaker equipment to allow paging to large areas, such as warehouses or loading docks. For installation instructions, see "Installing External Paging Equipment" on **page 1-331**.

IMPORTANT	When connecting external paging equipment to an LSM circuit, use a paging amplifier that provides talk battery. When connecting external paging equipment to the chassis, use a paging amplifier that *does not* provide talk battery.

NOTICE

Using a paging amplifier with talk battery on a phone may damage the phone or the module.

Power Cord Socket: For power supply requirements, see "Power Supply" on **page 1-61**.

USB Flash Drives

The USB-A serial port located on the front of the Mitel 5000 chassis receives a USB flash drive for backing up Unified Voice Messaging messages. For a current list of compatible USB flash drives, refer to Knowledge Base article 2468 on the Mitel Knowledge Base Center. For details on backing up voice mail messages, refer to the *MiVoice Office 250 Features and Programming Guide*.

Base Server Expansion Modules

This section describes the modules that can be installed in any of the four bays in the HX Controller chassis or any of the three bays in the CS Controller chassis to expand the capability of the MiVoice Office 250. Expansion modules include:

- T1/E1/PRI Module, 1-port – T1, T1/PRI, E1/PRI
  (PRI requires additional licensing)
- Dual T1/E1/PRI Module, 2-port – T1, T1/PRI, E1/PRI.
  (PRI and second port require additional licensing)
- Loop Start Module, 2-port – LSM-2 (no longer in production)
- Loop Start Module, 4-port – LSM-4.
- Basic Rate Module–2, 2-port – BRM-2.
- Digital Desktop Module, 16-port – DDM-16 (HX Controller chassis only)
- Single Line Module, 4-port – SLM-4.

Single and Dual T1/E1/PRI Modules

The T1M and T1M-2 modules support T1, T1/PRI, and E1/PRI spans. T1 refers to a digital method of transmitting voice and data. It is the 24-channel, time-division multiplex (TDM), 1.544 Mbps pulse code modulation (PCM) method used in the United States [Europe 2.048 Mbps].

The T1 standard, which was developed as a method of reducing cabling and increasing transmission quality between distant COs, is a subset of a multi-level Digital Signal (DS) scheme. T1 refers to the DS1 rate as transmitted across any type of media, such as copper wire, optical fiber, or microwave.

Since each T1 span actually consists of 24 individual circuits (or channels) multiplexed together, it is often less expensive to purchase a single T1 span than it is to purchase multiple individual trunks. Each of the 24 T1 channels can be programmed as a Loop Start, Ground Start, DID, E&M, or OPX (or as unequipped). Any combination is possible.

“Fractional” T1, often called FT1, is a special T1 service where only a portion of the 24 channels are used and paid for. To implement FT1 equip only those circuits that are provided.

To interface with the public network, each T1M and T1M-2 module is equipped with an on-board Channel Service Unit (CSU). Since the CSU is on-board, the programmer must select the line build-out (LBO) attenuation, which is available from the T1 provider.

To use an external CSU instead, the module can be set to DSX mode. However, the module and the external CSU must be located in the same building, and the cabling between them must be no longer than 133 ft.

When a T1 span connects the system to the public network or to another telephone system, the two phones must communicate at the same frequency. Since it is practically impossible for both phones to have clocks running at identical frequencies, one phone is given master clock status and the other phone is given slave clock status. The slave clock can then extract and lock-on to the master clock frequency from the incoming T1 data. This master–slave synchronization is programmed in the

database. When connected to the public network, a T1, T1/PRI or E1/PRI port should be designated as a public network slave.

When voice data is transmitted over a T1 span, a framing bit is added so that the receiving end can recognize where the information begins and ends, which is effective as long as the clock circuitry of the two phones is synchronized.

The system supports the two major types of framing schemes in use today: D4 Superframe, which is the most common, and Extended Superframe (ESF), which is required to support future ISDN and DMI features. For voice transmissions, D4 Superframe is adequate. ESF is more reliable and efficient for data transmissions.

Digital information is transmitted over the T1 span as a series of 1s and 0s (e.g., 1000 0000 0000 0001...). If the stream of digits contains too many consecutive zeroes, the T1 clock extraction circuitry will not operate correctly. To prevent this situation, T1 carrier equipment limits the number of consecutive zeros. The Mitel system supports the two major types of zero code suppression schemes in use today: Alternate Mark Inversion (AMI), also called bit-7 stuffing, and Bipolar Eight Zero Substitution (B8ZS). Again, for voice transmissions, AMI is adequate. However, for 64 Kbps clear channel ***data only*** transmissions, B8ZS is required.

Like the LBO attenuation, the framing scheme type and zero code suppression scheme type are also available from the T1 provider. In fact, when ordering a T1 span, always request the following information from the T1 provider:

- **Framing scheme**: D4 Superframe or ESF
- **Zero code suppression scheme**: AMI or B8ZS
- **Recommended line build-out attenuation**: 0 dB, 7.5 dB, 15 dB, or 22.5 dB

Each T1 port has circuitry to support up to 24 T1 span channels. The channels (or circuits) can be any combination of loop start trunks, ground start trunks, DID trunks, E&M trunks, or OPX circuits.

The T1M and T1M-2 modules are equipped with the following resources:

- **LED indicators**: Show the status of the module and circuits and to show T1 trunk alarms and module-level error occurrences. For additional information about LEDs, see the Diagnostics and Troubleshooting chapter in the *MiVoice Office 250 Features and Programming Guide*.
- **Digital span jack**: (RJ-48C; 8-pin, 8-conductor jack) Connects a T1, T1/PRI, E1/PRI span to the module. The T1M is equipped with one RJ-48C jack for the span and another RJ-48C jack to connect test equipment to monitor the span. The T1M-2 is also equipped with two RJ-48C jacks; however, both jacks are used for individual spans. The T1M-2 has no RJ-48C jack for test equipment.
- **DTMF transmitters**: *(T1M and T1M-2 modules only)* Generates DTMF tones for traffic on the T1M and T1M-2 modules. To use these transmitters, the **Enable On-Module Dialing** flag must be enabled in DB Programming. Refer to the *MiVoice Office 250 Features and Programming Guide*.

NOTE The T1M and T1M-2 module DTMF transmitters are *not* shared resources. Only the T1M and T1M-2 modules can use these transmitters.

Programming information that is *specific to each T1/PRI* installed in the system. For details, refer to the *MiVoice Office 250 Features and Programming Guide*:

- **Framing type**: The type of framing scheme, D4 Superframe or Extended Superframe (ESF), used by the T1 span connected to each module.

IMPORTANT	DB Programming will display an error message under the following circumstances: • Change the port type of one port if the other port is programmed (not None). • Change the Framing Type under a T1 port and the other port is programmed (not None). This field also exists under the T1/PRI port type, but it is OLM only. DB Programming displays: WARNING: Changes to this field are only recommended on an idle system as this may cause a short pause in audio on the other port of this module (but no dropped calls). Would you like to continue?

- **Zero code suppression scheme**: The type of zero suppression scheme, AMI (bit-7 stuffing) or Bipolar Eight Zero Substitution (B8ZS), used by the T1 span connected to each module.
- **Line build-out (LBO)**: The LBO attenuation of the T1 span connected to the module. This value can be determined automatically by the system, or it can be set according to the T1 provider's recommendation.
- **Loop Length**: (Applies only to T1 and T1/PRI modules.) This is displayed with a red "X" if the Haul Mode is set to Long Haul. It also provides optional loop lengths pertaining to a Short Haul configuration.
- **Haul Mode**: This applies only for a T1 or T1/PRI configuration (E1/PRIs do not display this field).
- **Reference clock programming**: All T1M, T1M-2, and BRM-2 modules and the motherboard must share the same clock signal so they can be synchronized with all the digital trunk interfaces (T1/PRI/BRI) connected to the system. This synchronization is accomplished by allowing one of the digital trunk interfaces to act as the system reference clock. The programmer must designate whether the digital trunk interface is a private network master, private network slave, or a public network slave. This clock-type assignment generates a prioritized list, which forces the public network slave interfaces to the top of the priority list. You can further prioritize which interface should be higher in the list, among interfaces of the same type.

  If more than one digital trunk interface is installed in the system, the software uses the Reference Clock list to determine a back up to the system reference clock. Then, if the module acting as the system reference clock fails or is taken out of service, the next module on the back-up priority list becomes the reference clock.
- **Error diagnostics**: If desired, the programmer can generate the current T1 or T1/PRI error counts, by hour and by day, for the selected T1 port. The error counts are displayed in tabular format.
- **Error thresholds**: The programmer can set numerical thresholds for certain T1 port errors in terms of the number of errors per hour and number of errors per day. When a threshold is exceeded, an error message is printed on the SMDR record.

The information that is programmed *specific to each circuit* (i.e., to each of the 24 channels) on the T1 module includes:

- **Circuit type**: Each individual circuit can be programmed as a loop start, ground start, DID, E&M, or OPX circuit.
- **DTMF or dial-pulse signaling**: When the system is in the default configuration, all trunk circuits are configured for DTMF signaling. If necessary, some or all of the circuits can be reprogrammed for dial-pulse signaling.

T1 circuits are assigned hardware address numbers according to the module's bay in the chassis and the location of the circuit on the module. For example, T1 circuit 3.1.21 is the 21st circuit on the T1M or T1M-2 module installed in Bay 3 of the chassis.

Special T1/E1 Cabling Requirements

Mitel recommends that special high-speed data cable with the following characteristics be used for all T1/E1 installations:

- 22 AWG (0.64 mm)
- Individually shielded two-pair (or with the transmit and receive pairs run in separate cables or binder groups)
- ABAM or PIC (a level-3 cable designed specifically for the T-carrier transmission rate)
- Maximum capacitance of 15 pF/ft
- Maximum attenuation of 5 dB/1000 ft (300 m) at 772 kHz and 70 °F (21 °C)

NOTE The programming prompts for LBO attenuation (and DSX mode, if used) assume that T1/E1 cabling possessing the above characteristics is being used. When transmitting over shorter distances, it may be possible to use standard two-pair voice frequency cable instead; however, the programmer may have to compensate by selecting a smaller LBO setting. Also, no matter which type of cable is used, the transmit and receive pairs must be shielded from each other or run in separate cables or binder groups.

T1M or T1M-2 Modules used for Private Networking

When using private networking to connect two nodes, each network span requires two T1M or T1M-2 modules, one for each end of the span. If using the Telco to connect the private networking span, order a clear channel, point-to-point T1 span from the service provider, for each networking span. Do not order premium PRI service, even though PRI will be used.

NOTE The Mitel system requires T1 channel 24 for the D-channel in any network span.

When a T1M or T1M-2 module is programmed for networking (Private Network switch type), the system automatically creates a node trunk group. All B channels residing on the same T1M or T1M-2 module are in the same node trunk group.

Two-Port Loop Start Module

Each two-port Loop Start Module (LSM-2) supports one or two analog loop start/CO trunks in the United States, and each quad Loop Start Module (LSM-4) supports 1–4 of the same type trunk. LSMs support the same number of local exchange trunks in Europe.

For optimal CO trunk transmission performance, each analog loop start trunk interface should be "hybrid balanced" in DB programming. Hybrid balancing refers to selecting the balance network that best matches the trunk's impedance. To determine an appropriate hybrid balance setting for analog loop start trunk interfaces, refer to the *MiVoice Office 250 Features and Programming Guide*.

LSM circuits incorporate a self-resetting fuse that should never need replacement. If the fuse should open momentarily—perhaps as the result of a lightning strike—it should reset within a few seconds.

NOTICE

LSMs contain no field-replaceable parts. If any insurmountable problem with an LSM should occur in the field, return the entire module to Mitel for testing and possible repair. Failure to follow this policy will void the warranty on the module.

Two green LEDs show the status of the LSM-2. The LED labeled ONLINE indicates if any circuit on the module is busy. The other LED, labeled REMOVE, lights up only when the system has been powered down and the condition is safe to remove the module from the bay. Also, when secured in the chassis bay, the metal front panel of the module connects to chassis ground to suppress electromagnetic interference (EMI).

When the system is in the default configuration, all trunks are configured for DTMF signaling. If necessary, some or all of the trunks can be reprogrammed for dial-pulse signaling through DB Programming.

NOTE Single line phones generate their own DTMF tones when dialing. If used on dial-pulse trunks, both the DTMF tones generated by the phone and the dial-pulse signals generated by the system are sent to the CO, which may result in a problem if the CO recognizes both. To avoid this problem, single line sets should be restricted to ARS-only when using dial-pulse trunks. Then, only the ARS-generated dial-pulse signals will be sent to the CO.

CO circuits are assigned hardware address numbers according to the module's bay in the chassis and the location of the circuit on the module. For example, when using the initial configuration (see paragraph 3.91), CO circuit 02.02.01 is the second circuit on the LSM-2 or LSM-4 installed in bay 2. Each trunk module circuit is assigned a six-digit circuit number XX.YY.ZZ, where XX = the module bay location (01–03), YY = the circuit number on the module (01 or 02), and ZZ = whether the circuit has more than one purpose (currently, 01 only).

When the system is in the default configuration, the first four buttons on each phone are assigned as CALL buttons for incoming call indication/access. The ANSWER button may also be used to access incoming calls. No individual trunk buttons or trunk group buttons are assigned. Button assignments can be designated in DB Programming. For more information, refer to the *MiVoice Office 250 Features and Programming Guide*.

Four-Port Loop Start Module

The four-port loop start module (LSM-4) supports four CO [local exchange] trunks. The LSM-4 functions the same as the LSM-2. For complete installation instructions, see "Installing a Four-Port Loop Start Module (LSM-4)" on **page 1-190**. For programming refer to *MiVoice Office 250 Features and Programming Guide*.

Basic Rate Module

IMPORTANT Basic Rate Interface (BRI) capability is supported only in Mitel Europe installations.

The Basic Rate Module (BRM-2) provides Basic Rate 2 interface ISDN service for communication and is available only in the European market. Each of the two BRI ports consists of two bearer (B) channels and one data (D) channel with High-level Data Link Control (HDLC) support. Each BRM-2 module in the chassis supports a total of four BRI trunks.

The ports provide full I.430 ITU S/T ISDN support for trunks (TE mode), but does not support stations (NT) mode. The BRM-2 does not support video conferencing or BRI devices and phones.

16-Port Digital Desktop Module (DDM-16)

Up to 16 digital devices can be connected to the DDM-16, and there can be up to three DDM-16s installed in any of the four HX Controller expansion bays. The same digital telephones and SLAs supported on the DEI's DEM-16 are supported on the DDM-16. Each DDM-16 includes:

- A single 48-pin connector and a special DDM cable for connection to a station block at the Main Distribution Frame
- Status LEDs (online, busy, and remove)
- Module release lever for safely removing the module from the bay
- DSP resources (4 DTMF receivers, 5 tone generators, and 16 speakerphones for devices connected to the module)

NOTE This module functions only in the HX Controller chassis. Do ***not*** attempt to install it in the CS Controller chassis, as damage may occur. Also, DDM-16s are not subject to any of the voice channel trade-offs that the DEM-16s are subject to when used in two-DEI or four-DEI mode.

Four-Port Single Line Module Characteristics

Up to four SLM-4s can be installed in the HX Controller chassis, but only one SLM-4 can be installed in the CS Controller chassis.

The Single Line (SL) application installs as a part of the MiVoice Office 250 Version–Release suite of software packages, along with other application packages that support SLM-4 functionality, such as DB Programming, Call Processing, etc. The SLM-4:

- Is hot-swappable.
- Is designed to be inserted into one of the available bays on the back side of the Mitel 5000. The form factor used for the SLM-4 is the same as used for the T1M or T1M-2, LSM-2, LSM-4, and BRM-2 modules.
- Supports two green status LEDs. One shows ONLINE status, and the other shows the ready to REMOVE status. The SLM-4 provides two RJ-14 jacks, each wired to USOC T568A specifications. Together, the two jacks support four single line analog phones.
- Does ***not*** require software feature licensing.
- Does ***not*** operate on a v1.x system. If you plug an SLM-4 into a v1.x system, a warning message appears and Message Print indicates the board will not come online.
- Does ***not*** support dial pulse dialing.
- Does ***not*** respond to the Auto Equip feature—SLM-4 modules must be programmed manually in DB Programming. For instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

Processing Server (PS-1)

This section describes specifications relevant to the optional PS-1/vPS-1. See the MiVoice Office 250 Enginerring Guidelines for details on when a PS-1/vPS-1 might be needed.

The physical PS-1 is a rack-mountable Mitel Server Applicance, using the Dell Power Edge R210 Server. For specifications and characteristics please refer to www.dell.com.

The vPS-1 is a virtual application (vApp) which is deployed through vCenter Server with the vSphere Client. The PS-1 vApp runs on vSphere 5.0, 5.1, or 5.5. For a list of servers that are approved for use with product releases 5.0, 5.1 and 5.5, consult the VMware Compatibility Guide (http://www.vmware.com/resources/compatibility/).

Processing Server Characteristics (Physical PS-1)

Table 12 summarizes the physical, electrical, and server computing characteristics of the physical PS-1. For a summary of suitable environmental characteristics see “Environmental Requirements” on page 60.

Table 12: PS-1 Specifications

Characteristic	Description
Physical dimensions	
Height	1.67 in. (4.26 cm)
Width	17.1 in. (43.1 cm)
Depth	15.5 in. (39.4 cm)
Weight	15 lb (6.8 kg), approximately
Electrical	
Power requirements	90–264 VAC, 47–63 Hz (USA); 240 VAC, 50 Hz [Europe]
Server computer	
Processor	Intel® Core i3 Class (2 GHz minimum)
Operating System	Linux® Debian v2.6
Hard drive capacity	Non-redundant 250GB SATA
RAM	8 GB ECC DDR3
System interfaces	
LAN	Dual Port Embedded Gigabyte Ethernet
Online Monitor (OLM)	RS-232, DB-9 serial port with null modem cable
Security Key	USB 2.0

PS-1 BIOS Settings

Table 13 lists the required BIOS settings for the Dell PowerEdge R210 Rack-mount server.

Table 13: Dell PowerEdge R210 BIOS Settings

BIOS Setting	Required Settings
Main Menu => System Time	Set to site time zone
Main Menu => System Date	Set to current date
Main Menu => Report Keyboard Errors	Report
Memory Settings => System Memory Testing	Enabled

(Sheet 1 of 2)

Table 13: Dell PowerEdge R210 BIOS Settings

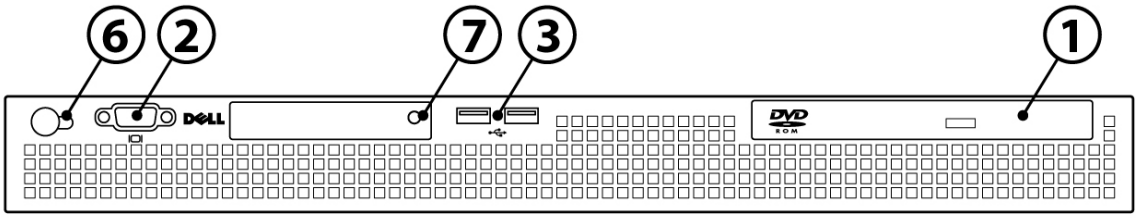
BIOS Setting	Required Settings
Processor Settings => Logical Processor	Enabled
Processor Settings => Virtualization Technology	Disabled
Processor Settings => C States	Disabled
SATA Settings => SATA Controller	AHCI
Boot Settings => Boot Mode	BIOS
Boot Settings => Boot Sequence Retry	Enabled
Integrated Devices => Embedded NIC1 and NIC2	Enabled
Integrated Devices => OS Watchdog Timer	Enabled
Power Management => Power Management	Maximum Performance
System Security => Power Button	Enabled
System Security => NMI Button	Disabled
System Security => AC Power Recovery	Last
System Security => AC Power Recovery Delay	Immediate
	(Sheet 2 of 2)

PS-1 Connection Interfaces

This section provides front and rear view representations of the Dell PS-1 chassis.

NOTE The On/Off switch on the front of the chassis controls power to the system, and it should remain On at all times.

Figure 3: Processing Server (PS-1) Chassis, Front View



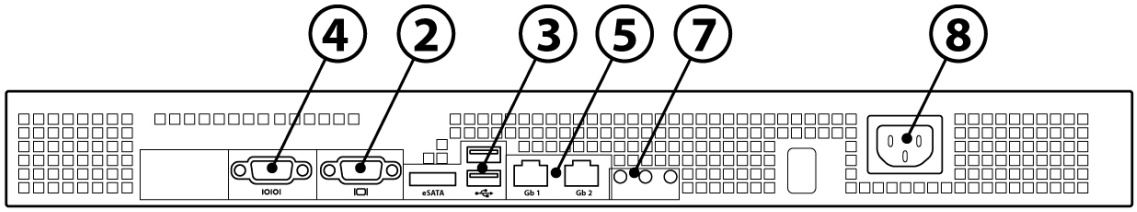
Item	Description
1	Digital Video Disk (DVD) drive. (Use for initial installation and system recovery.)
2	Video port. (DB-15 VGA connection to monitor. Can use front or rear ports.)
3	USB ports. The USB security key can be used in any USB port. (Use also to connect keyboard and mouse if necessary.)
6	Power LED.
7	System Identification LED. (Used for system location. See www.dell.com for details.)

NOTICE

Accidental System Reset. Do not mount the PS-1 in a rack directly above the Base Server. If you do, pulling down the front cover (not shown) of the PS-1 may cause the edge of the cover to press the three middle buttons on the LCD panel controls, causing the system to reset. If you mount the PS-1 in the same rack as the Base Server, make sure the hinged front cover of the does *not* open onto the front of the Base Server. Mitel recommends mounting the PS-1 below or away from the Base Server and Digital Expansion Interface (DEI) units.

Figure 4 shows the back of the PS-1 unit.

Figure 4: Processing Server (PS-1) Chassis, Back View



Item	Description
2	Video port. (DB-15 VGA connection to monitor. Can use front or rear ports.)
3	USB ports. The USB security key can be used in any USB port. (Use also to connect keyboard and mouse if necessary.)
4	Serial port. (Laptop connection using RS-232 cable/DB9M to DB9F straight-through connector, part number 813.1683.)
5	Ethernet ports.
7	System Identification LED. (Used for system location. See www.dell.com for details.)
8	Power cord plug.

Digital Expansion Interface

This section describes the specifications of the optional Digital Expansion Interface (DEI) unit and the modules that it supports—the Digital Endpoint Module (DEM-16) and the eight-port Single Line Module (SLM-8).

DEI Unit

Physical dimensions of the DEI unit appear in **Table 14**.

Table 14: DEI Physical Dimensions

Dimension	Measurement (Metric)
Length	14.50 in. (36.83 cm)
Width	17.00 in. (43.18 cm)
Height	01.75 in. (4.45 cm) (1 U)
Weight	09.00 lb. (4.08 kg)

A green power (PWR) LED on the front of the chassis indicates whether the DEI is plugged-in to an active electrical circuit. Also, three sets of LEDs show the status of the three module bays. The active status of each bay is shown by a green online (ON) LED and a yellow BUSY LED. Three bays at the back of the chassis accept DEM-16 modules, and the following connection interfaces are provided:

- The system power cord socket with built-in fuse holder (fuse is 250 V, 2 A). Power supply requirements appear in “Power Supply“ on **page 61**.
- Two DB-15 connectors. The 15-pin D-subminiature socket connectors provide electrical interface to the Mitel 5000 chassis and to either another DEI or an optional, external Enterprise Messaging system.
- An earth ground lug.

Digital Endpoint Module

The Digital Endpoint Module (DEM-16) supports 16 digital telephones, and three DEM-16s can be installed in a DEI.

The status for each DEM-16 is shown with LEDs on the faceplate of the module. When lit, a yellow LED indicates that at least one circuit on the DEM-16 is BUSY, as printed on the module. When lit, a green LED indicates the module is ONLINE. The online/Off-line status for each module also appears on the Administrative Web Interface.

DEM-16 ports are programmed in DB Programming. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Eight-Port Single Line Module Characteristics

Up to three SLM-8 modules can be installed in each Mitel Digital Expansion Interface (DEI) unit attached to the HX Controller or CS Controller chassis.

The SLM-8:

- Operates only on systems running v2.2 or later system software.
- Is hot swappable.
- Is designed to be inserted into one of the three bays of the MiVoice Office 250 Digital Expansion Interface (DEI) chassis. The form factor used for the SLM-8 is the same as used for the Digital Endpoint Module (DEM-16) module.
- Supports two green status LEDs. One shows ONLINE status, and the other shows the ready to REMOVE status. The SLM-8 provides four RJ-14 jacks, each wired to USOC T568A specifications. Together, the jacks support eight single line analog phones.
- Does ***not*** require software feature licensing.
- Does ***not*** support dial pulse dialing.

Audio Saturation Limit for Single Line Ports

If the audio power level feeding into a single line port exceeds the industry standard, distorted audio may be heard on the conversation. To the listener, the distortion may be an unexpected burst of noise that repeats.

This unusual event may occur on any single line voice port supported by v2.1 or later. The phone may be connected to a Single Line Module (SLM-4 or -8) or to either of the single line ports built-in to the Mitel 5000. Distortion-related factors include the makes and models of phones being used and the volume level of the speaker. The root cause of the problem has been traced to the analog single line interfaces between the MiVoice Office 250 and phones that transmit at abnormally high power levels.

NOTICE

Audio Distortion Possible. Distorted audio may occur with various analog single line phones if audio input volume exceeds the industry standard.

For troubleshooting and corrective action suggestions, refer to the *MiVoice Office 250 Features and Programming Guide*.

Auto NAT Detection and Digital Equipment

This section describes the software version requirements and phones that do and do not support Auto NAT Detection.

Software Version Requirements

For the Auto NAT Detection feature to function, the system must be running Call Processing v1.2 or later.

Phones

With applicable licensing, Auto NAT Detection supports multi-protocol phones 8600, 8620, 8622, 8662, and 8690; 8660, IP PhonePlus, and IP Single-Line Adapter (IP SLA); and, the 8602 IP softphone.

Auto NAT Detection does not support the following phones. The NAT setting must be set manually for:

- Wireless IP phones 8664, 8665, and 8668.
- IP gateway trunks, neither Station Initiation Protocol (SIP) nor Media Gateway Control Protocol (MGCP).
- SIP phones.
- Audio and Web Conferencing (AWC)

52xx/53xx phones do not support the Automatic Network Address Translation (NAT) address type. In DB Programming, you must select either Native or NAT for private and public networks. Refer to the *MiVoice Office 250 Features and Programming Guide*, part number 580.8006, to configure the NAT address type.

System Resources with DEI

Installing up to four DEI units in a MiVoice Office 250 increases the number of resources available for each of three DEM-16s.

- In 2-DEI mode, the system has the same functionality as pre-v3.x except for the on-module resources. You can equip up to six DEMs on two DEIs. Each DEI has 96 timeslots available for 48 digital telephones and each DEI also provides additional features such as OHVA, dual-SLAs, and MDPMs. Each DEM provides four DTMF receivers.
- In 4-DEI mode, there are 48 voice channels to each DEI; this provides one voice channel per digital telephone. Features that require a second voice channel, such as OHVA, MDPM, and dual-SLA, require special attention. Enable these features using the current methods; however, these features use a voice channel that could otherwise be used by a telephone on the same DEI. DB Programming counts the time slots used by these features and enforces the maximum timeslot limit for each DEI.
- Features that require a second voice channel, dual-SLAs, OHVA, and MDPMs are affected when expanding DEI capacity. When in 4-DEI mode, do not overallocate a DEI with these features. Enabling features that require a second voice channel will likely reduce the maximum phone capacity in 4-DEI mode.
- The DEM four DTMF receivers, eight caller-ID transmitters, and four caller-ID receiver DSP resources will be disabled in 4-DEI mode. The DEM 16 speakerphone resources will still be available to digital keysets connected to the DEM in this mode. The loss of the caller-ID transmitters and receivers are offset by using the DSP for these functions. The four DTMF receivers on each module are offset because the Dual T1/E1/PRI Module has DTMF receivers built into each channel.

A DEI is “slightly over allocated” if it has more than 48 time slots programmed and the system is in 4-DEI Mode. It is “extremely over allocated” if it has more than 53 time slots programmed and the system is in 4-DEI Mode. DB Programming should (in most cases) prevent the system from ever becoming extremely over allocated.

When programming, there are two basic cases where a warning or error message related to time shot usage may appear:

- If the time slot values for a DEI are between the overallocated (53 time slots) and allowed (48 time slots), you are notified that you have entered a '”blocking scenario” and you can choose whether to continue changing the time slot usage.

- When the DEI has reached the overallocated value. In this case, you are warned that you have reached the time slot limit and you can not allocate any other time slots until you have unallocated one or more of the time slots currently allocated.

Cabling for DEI Digital Capability

A 15 ft (5 m.) 25-pair cable is available for connecting DEM-16 ports to the main distribution frame (MDF). The cable, part number 813.1814, is terminated on one end with four 8-pin, 8-conductor plugs wired to RJ-45 specifications using the 568B standard. The other end of the cable is terminated with a male Amphenol-type connector. The cable is designed for quick connection of digital telephones that have been terminated on a female Amphenol-type connector at the MDF. For usage instructions, see “Connecting Station Cables to the MDF” on **page 1-233**.

For the HX Controller DDM-16 station cabling information, see “Cabling to Phones” on **page 1-65**.

Unified Voice Messaging (UVM)

This section describes the measurable parameters of the Unified Voice Messaging feature, including voice mail storage limits, ports available for licensing on the Communication Servers, voice mail recording process, and authentication supported.

With v5.0, the system supports 24 or 32 licensed UVM ports, as shown in **Table 15** below.

The ports are allocated dynamically up to the number of valid UVM licenses. Ports are licensed four at a time and may *not* be licensed individually. If the system is equipped with a PS-1vPS-1, UVM is hosted on the PS-1/vPS-1.

Table 15: Maximum Number of Supported UVM Ports

Controller	Standalone	PS-1/vPS-1
CS	24	32
HX	32	32

Table 16 shows the relationship between the number of valid UVM licenses and the maximum number of active UVM ports in a system. DBP does *not* allow dynamic allocation of more UVM ports than are licensed for the system and that have been manually programmed into the appropriate Time Slot Group.

Table 16: UVM Licenses, Time Slot Group Ports, and IP Resources Allocated

Valid UVM Mail Port Licenses	Ports Defined in Time Slot Group	IP Resources Dynamically Allocated	Active UVM Voice Mail Ports
0	0	0	
4	4	0–4	
8	8	0–8	
12	12	0–12	
16	16	0–16	
20	20	0–20	
24	24	0–24	
28	28	0–28	
32	32	0–32	

If all of the IP resources available for UVM are in use, the next caller is camped-on to wait for a resource to become available. For additional information, refer to the Voice Processing chapter in the *MiVoice Office 250 Features and Programming Guide*.

UVM Voice Mail Ports

Having up to 32 UVM ports available on the system provides increased availability and response time for voice processor applications. Under normal business conditions, increased ports allow fast response to applications such as voice mail, Message Notification/Retrieval, Automated Attendant, and Record-A-Call. Programmers can reserve up to the maximum ports supported by the license installed. For programming instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

The system supports up to 32 UVM ports on the PS-1. When the PS-1 is installed, the eight-port UVM feature is disabled. The PS-1 initially provides four UVM ports and additional ports can be

licensed and added in four-port increments. For programming instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

UVM Forward to E-Mail Characteristics

UVM Forward to E-Mail possesses the following characteristics:

- Forward to e-mail for SMTP–MIME servers only.
- Forward voice mail as **.wav** and **MP3** file attachments in e-mail messages.
- The **.wav** encoding format plays on a standard PC running Windows®.

Initial Recording Handling

If **Forward and Copy** is selected as the e-mail forwarding type in DB Programming, the initial recording is left at the voice mailbox until deleted using voice mail.

> **NOTE** If both Remote Messaging and Forward to E-Mail features are enabled, the Forward and Copy option must be selected. If the Forward Only option is selected, the user will not receive Remote Messaging notifications.

If **Forward** is selected at the voice mailbox as the e-mail forwarding type, the initial recording is deleted. Use this setting with caution because e-mail delivery on the Internet is not guaranteed. Only one copy is kept on the system of the initial recording.

Temporary Files

Temporary files are used during the conversion from the initial recorded message to the SMTP–MIME format. The temporary storage required may impact overall UVM storage capacity.

Authentication Supported by UVM

Users do not need to configure anything for authentication beyond setting their username and password. However, the SMTP mail software package supports three types of authentication that occur automatically when connecting with the SMTP server. The three types of authentication are:

- PLAIN
- LOGIN
- CRAM-MD5

This package does not support Microsoft SPA (Secure Password Authentication), also known as NTLM. If the server supports CRAM-MD5, this is the most secure method supported.

Many SMTP servers validate messages to prevent what is referred to as an “open relay.” These SMTP servers require that the source and destination addresses are valid e-mail domains. Improper configuration of the settings within the MiVoice Office 250 may result in rejected e-mail messages.

End-users may also use anti-spam software to filter out unwanted e-mail messages. End-users may need to configure their anti-spam software to allow the forwarded voice mail to reach the user's in-box.

Trunking

The MiVoice Office 250 supports various combinations of trunking across T1, T1/PRI, and E1/PRI spans, IP gateway trunks, and loop start trunks.

Trunking Capacity and Constraints

The MiVoice Office 250 supports up to 104 trunk groups in DB Programming and each trunk group supports a variable number of CO and IP gateway trunks. Up to 208 trunks can be programmed per trunk group in DB Programming, but the actual number of trunks that can be connected depends on the physical constraints of the system. For capacity, see “System Capacities” on page 67. For programming, refer to the *MiVoice Office 250 Features and Programming Guide*.

For hard-wired, non-IP gateway trunking, system capacity is determined by the number of bays equipped with T1M, T1M-2, or LSM-2 modules. The system provides three bays (CS Controller) or four bays (HX Controller) for multi-trunk modules in addition to the loop start trunk circuits built-in to the back of the Base Server.

For software-licensed IP gateway trunking using SIP gateway trunks, actual capacity is determined by the number of IP resources and the installation of enabling hardware. A PEC-1 can be installed to provide additional IP resources. Based on requirements, the customer provides appropriate equipment to enable IP gateway trunking. For a detailed description of the hardware and procedures required for installation of SIP or MGCP IP gateway trunks, see “Installing SIP Gateways” on **page 1-337**.

Hard-Wired (Non-IP Gateway) Trunking

In an all-trunk system equipped with required devices and licensing, the MiVoice Office 250 could support from 2 to 244 hard-wired, non-IP gateway trunks depending on the chassis type. At the low end of the range, the system could operate on the two loop start trunk circuits that are built-in on the CS Controller chassis. No trunk modules would be installed in the bays. At the high end of the range, 244 trunks would be available in an HX Controller system equipped with eight E1/PRI spans and four loop start trunks.

Capacities of All-Trunk systems

Table 17 summarizes system trunk capacity in terms of the trunk module type and the possible number of trunks provided in all-trunk systems. A typical installation may include a combination of hard-wire trunks and IP gateway trunks. See also Table 8, “Maximum Licensing Capacities,” on **page 71** for IP capacities.

Table 17: Trunking Capacity by Type of Module

Module Type	Trunk/Span Type	Modules (CS/HX)	Ports or Channels per Module	Trunk Capacity (CS/HX)	Built-In LS Ports (CS/HX)	Total Trunks (CS/HX)
T1M[a]	E1/PRI	3/4	30	90/120	2/4	92/124
T1M-2	E1/PRI	3/4	30	180/240	2/4	182/244
T1M[a]	T1	3/4	24	72/96	2/4	74/100
T1M-2	T1	3/4	24	144/192	2/4	146/196
T1M[a]	T1/PRI	3/4	23	69/92	2/4	71/96

Table 17: Trunking Capacity by Type of Module (Continued)

Module Type	Trunk/Span Type	Modules (CS/HX)	Ports or Channels per Module	Trunk Capacity (CS/HX)	Built-In LS Ports (CS/HX)	Total Trunks (CS/HX)
T1M-2	T1/PRI	3/4	23	138/184	2/4	140/188
LSM-2[a]	Loop Start	3/4	2	6/8	2/4	8/12
LSM-4	Loop Start	3/4	4	12/16	2/4	14/20
N/A	IP Gateway Trunks[b]	0	N/A	32[c]	2/4	34

a. No longer in production.
b. Requires licensing and installation of enabling hardware.
c. Lab-tested to 32 trunks. Actual capacity depends on the availability of IP resources, the vocoder used, and the system configuration.

SIP Trunk Capacities

The MiVoice Office 250 can be equipped with up 100 SIP trunks, although a PEC-1 is recommended above 25. Actual capacity depends on the availability of IP resources, the vocoder used, and the system configuration.

Phones

A variety of multi-protocol, IP-only, and digital telephones can be used on the MiVoice Office 250. The combination of phones depends on the number and type of phone modules installed. For phone illustrations, see “Phones“ on **page 479**. For end user feature descriptions, refer to the End User Features chapter in the *MiVoice Office 250 Features and Programming Guide*.

At least one display phone should be installed to allow an administrator the capability of seeing system alarm messages, program system data, and receive unsupervised CO recalls. When the system is in the default configuration, extension number 1000 is preassigned as an Administrator and as the Primary Attendant. If extension number 1000 is deleted from the database, an alternative extension must be assigned as the Administrator and the Primary Attendant.

NOTE

Optional Headsets

A headset may be attached to phones by unplugging the handset from the handset jack on the base of the phone, plugging the headset into the handset jack, and entering a feature code to enable the headset. The **Speaker** button, which is used to turn the headset on and off, is lit when placing and receiving calls and unlit when the headset is not in use. The phones are compatible with industry-standard, 4-conductor, modular headsets that have electret microphones. For installation instructions, refer to the Installation chapter in this manual.

Some of the phones have dedicated headset jacks on the rear of the phone. This allows the phone to have a headset and handset plugged in simultaneously. The headset, however, must still be enabled using the feature code. The Speaker button is used to turn the headset on and off and to indicate when the headset is not in use. In addition, the hookswitch is enabled even when the headset is turned on. This allows the user to answer a call with the handset without having to disable the headset.

Multi-Protocol Phones

The following multi-protocol phones support the full functionality IP mode but not the functionality of the Session Initiation Protocol (SIP) mode, on the MiVoice Office 250 platform:

- 8600 – No Display
- 8620/8622 – With 2-Line Display
- 8662 – With 6-Line Display
- 8690 – With Multimedia Touchscreen

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform. See page 275 for changing from SIP mode to IP mode.

IP Phones

The MiVoice Office 250 supports the following IP-only phones:

- 52xx/53xx-series phones (5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, and 5360)
- 8660 – 6-Line Display
- 8664 – 2-Line Display 802.11b Wireless
- 8665 – 2-Line Display 802.11b Wireless

- 8668 – 2-Line Display 802.11b Wireless

52xx/53xx Phone Specifications

Table 18 shows some of the 52xx/53xx phone specifications. For the latest specifications for all phones (including the newer 5304, 5312, 5324, 5320, and 5360 phones) see their respective data sheets on Mitel Online (Products / Desktops / IP Desktop / Related Links).

Table 18: 52xx/53xx Phone Specifications[a]

Specification	5212	5224	5330	5340
Size (L x W x H)	9.5” x 8” x 4.5” (24 cm x 20.5 cm x 12 cm)			
Ports	2			
Power	**Output**: 48VDC 250 mA **Input**: 100V to 240V 50 to 60 Hz **LAN Power**: 802.3af PoE	**Output**: 48VDC 250 mA 24VDC 1.7A (reserved for future use) **Input:** 100V to 240V 50 to 60 Hz **LAN Power**: 802.3af PoE	**Output**: 48VDC 250 mA 802.3af PoE **Input**: 100V to 240V 50 to 60 Hz **LAN Power**: 802.3af PoE	
Power Consumption	3.9 W			4.8 W
Processor Type	ARM946E-S		MIPS	
Processor Speed	140 MHz		165 MHz	
DSPs	N/A	1		
DSP Speed	N/A	74 MHz	125 MHz	
Silicon (Processor)	0.18µ CMOS		0.13µ CMOS	
Codec/Wideband	N/A		G.722 (currently not supported)	
RAM	8 MB		16 MB	
ROM	2 MB		8 MB	
Voice Traffic	RTP over UDP			
Signaling	MiNet over TCP			
Ethernet/Voice QoS	IEEE 802.1p/q			
Weight	2.8 lbs. (1.3 kg)			
Mean Time Before Failures (MTBF)	10 years			
Codec/Compression	G.711, G.729a			

a. Not all features are supported across all IP platforms.

52xx/53xx Phone Environmental Conditions

Table 19 shows recommended 52xx/53xx phone environmental conditions.

Table 19: Environmental Conditions

Temperature	Humidity
Operational: 40°F to 120°F (4°C to 49°C) Storage: –40°F to 150°F (–40°C to 66°C)	95% (non-condensing) 15% (non-condensing)

52xx/53xx Phone Regulatory Standards

Table 20 shows Mitel phone electromagnetic compatibility (EMC) and safety regulatory standards.

Table 20: EMC and Safety Regulatory Standards

Standard	Canada	USA	EU
EMC	ICES-003 (Class A)	CFR Title 47 Part 15 (Class A)	RTTE Directive (1999/5/EC) EN55024 EN55022
Safety	CAN/CSA/ CSA-C22.2 No. 60950-01	UL60950-01	RTTE Directive (1999/5/EC) EN60950-01

Support for the 5360 IP Phone

The 5360 IP phone (part number 50005991) has a seven-inch (17.8 cm), backlit, high-resolution (800 x 480), color, touch-screen display with brightness controls. For one-touch feature access, the 5360 has eight context-sensitive softkeys and 42 self-labeling programmable keys (three pages of 14 each). It includes an embedded screen saver application and text size control.

The Gadget Sidebar provides one-touch access to embedded applications (such as turning the display's backlight on or off). And, a 10/100/1000 auto-sensing Gigabit Ethernet LAN connection allows unconstrained bandwidth from the network to the desktop. The phone supports the Mitel Cordless Handset and Headset, the Mitel Line Interface Module, the Mitel 5310 IP Conference Unit, and the Mitel Wireless LAN Stand.

The 5360 IP phone requires MiVoice Office 250 v4.0 or later system software. It is not backward compatible with earlier versions of system software. The 5360 IP phone requires a Category D license.

5360 IP Phone Features

The 5360 IP phone has the same basic functionality as the 5340 IP phone except that the 5360 IP phone:

- has a color display.
- does not have "physical" programmable buttons
- has "virtual" programmable buttons accessed via a touch-screen display
- has LED-like button statuses represented via graphics on the display
- has an eight-button softkey region (four on the left and four on the right)
- stores brightness/contrast settings in the phone (not in the system database)
- has a Gadget Sidebar for launching applications

- has a backlight off unction (to turn the display off if desired)
- has a calibrate display function (to re-calibrate the touch screen accuracy if necessary)

For additional information about the phone and available system features, refer to the *Mitel 5360 IP Phone User Guide*, part number 550.8124.

Specifications

For information on the 5360 IP phone specifications, access the Mitel Online (MOL) Web site and navigate to the 5360 IP Phone Web page (http://tinyurl.com/yhjdcqk). Click the data sheet link at the bottom of the page. You will need your MOL user name and password.

Support for the 5320 IP Phone

The 5320 IP phone (part number 50006191) has a 160 x 320 (pixels) liquid crystal display (LCD), a variety of feature buttons, on-hook dialing, and a Ring/Message Indicator. For one-touch feature access, the 5320 has a number of self-labeling programmable buttons. It includes an embedded screen saver application and text size control (see page 106 for details).

The 5320 IP phone requires MiVoice Office 250 v3.2 or later system software. It is not backward compatible with earlier versions of system software. The 5320 IP phone requires a Category D license.

For additional information about the phone and available system features, refer to the *Mitel 5320/5330/5340 IP Phone User Guide*, part number 550.8123.

Unsupported Features

The 5320 IP phone has the same basic functionality as the 5330 IP phone except that:

- the top three buttons do not have LEDs.
- there is no backlit display.
- it does not support a cordless module or cordless devices.
- it does not support multiple pages of programmable keys.

Specifications

For information on the 5320 IP phone specifications, access the Mitel Online (MOL) Web site and navigate to the 5320 IP Phone Web page (http://tinyurl.com/mcysgu). Click the data sheet link at the bottom of the page. You will need your MOL logon user name and password.

Support for the 5304 IP Phone

The 5304 IP phone (part number 51011571) is a two-line IP phone. It has a 20-character two-line backlit liquid crystal display (LCD) screen. The 5304 IP phone provides a number of programmable keys for one-touch feature access and one designated Call key.

The 5304 IP phone requires MiVoice Office 250 v3.2 or later system software. It is not backward compatible with earlier versions of system software. The 5304 IP phone requires a Category D license.

For additional information about the phone and available system features, refer to the *Mitel 5304 IP Phone User Guide*, part number 550.8034.

Unsupported Features

The 5304 IP phone has the same basic functionality as the 5212 IP phone except that the 5304 does not have an external microphone. The 5304 IP phone does not support the following features and peripherals:

- On-hook dialing
- Handsfree operation
- Dedicated headset port

> **NOTE** Although the 5304 does not include a dedicated headset port, users can use a headset on the phone. However, when headset mode is enabled, users require a speaker button to connect to and disconnect from calls.

- Mute button
- Speaker button

> **NOTE** The 5304 keymap can be reprogrammed to provide headset users with a Speaker button.

- Dual-color LEDs. The System flag “Use Green LEDs for Direct Rings” does not apply.
- Off-Hook Voice Announce
- Cordless support (cordless module, cordless handset, cordless headset)
- Wireless Local Area Network (LAN) Stand
- Gigabit Ethernet Stand

Specifications

For information on the 5304 IP phone specifications, access the Mitel Online (MOL) Web site and navigate to the 5304 IP Phone Web page (http://tinyurl.com/dz6m95). Click the data sheet link at the bottom of the page. You will need your MOL logon user name and password.

Enhanced Support for 5312/5324 IP Phones

Prior to v3.2, the 5312/5324 IP phones were supported by the MiVoice Office 250 software; however, when the phones came online, they simulated the 5212/5224 IP phones.

With v3.2 and later, the 5312/5324 IP phones now come online as the appropriate model IP phone.

The 5312/5324 IP phones are designed to replace the 5212/5224 IP phones. The differences between the phones are as follows:

- The 5312 provides full-duplex audio in handsfree mode versus the half-duplex audio provided by the 5212 in handsfree mode.
- Both the 5324 and 5224 IP phones support the 12-key and 48-key PKMs. However, the 5324 phone has an updated PKM module on the back of the phone to connect to the PKM.
- The software loads for the 5312/5324 IP phones are stored differently than the 5212/5224:

- For 5312/5324 phones, the main software load is stored in Flash memory. Therefore, these IP phones are capable of running in an environment without a TFTP server.
- For the 5212/5224 phones, the main software load is stored on the TFTP server. Therefore, these phones require an environment with a TFTP server.

For additional information about the phones and available system features, refer to the *Mitel 5312/5324 IP Phone User Guide*, part number 550.8035.

Specifications

For information on the 5312/5324 IP phone specifications, access the Mitel Online (MOL) Web site and navigate to the 5312 IP Phone Web page (http://tinyurl.com/lv348q) or the 5324 IP Phone Web page (http://tinyurl.com/mbnlks). Click the data sheet link at the bottom of the page. You will need your MOL logon user name and password.

5330/5340 IP Phone Display Enhancements

With v3.2 and later, the MiVoice Office 250 supports two display enhancements for the 5330 and 5340 IP phones:

- **Embedded screen saver application**: 5330 and 5340 IP phones now include an embedded screen saver that users can configure for their display. Configuration options include the screen saver timeout, display image, and brightness options.
- **Text size control enhancement**: 5330 and 5340 IP phone users can now control the size of the text that appears on the display for programmable buttons. Text size options include:
  - *Large text*: Provides one line of basic information for the features and functions programmed for the button. Large text is enabled by default.
  - *Small text*: Provides a first line of basic information and a second line of supplemental information for the features and functions programmed for the button.

  The text size control enhancement does not apply to the text size that appears on the display for the menu buttons.

  The table below includes several examples for the large text and small text display options.

Programmable Button	Large Text Example	Small Text Example	Second line provides...
Station Speed Dial	Monica S	Monica S 25598	The station speed-dial number programmed for the button.
System Speed Dial	Chandler TS	Chandler TS 4809619000	The system speed-dial number programmed for the button.
Direct Station Selection (DSS)	Jessica A	Jessica A 28899	The extension number programmed for the user.
Programmable Feature	DND	DND (372)	The feature code that corresponds to the feature programmed for the button.
Intercom Call (IC)	IC	IC Charlie B	The name or number for the other party on the internal call.
Call Key	480-961-9000	480-961-9000 Mitel Networks	The caller ID (if available).

For additional information about the phone and available system features, refer to the *Mitel 5320/5330/5340 IP Phone User Guide*, part number 550.8123.

NOTE These display enhancements to the 5330/5340 IP phones are included as standard features with the 5320 IP phones.

52xx/53xx and 86xx Phone Differences

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform. See page 275 for changing from SIP mode to IP mode.

The following are differences in behavior between 52xx/53xx IP phones and 86xx multi-protocol phones.

- When using a 52xx/53xx phone, you cannot use the handset or speakerphone when the phone is in Headset Mode. 86xx phones allow you to use the handset or speakerphone when the phone is in Headset Mode.
- 52xx/53xx phones automatically save volume levels (you do not have to press the center of the Volume button to save volume levels as with 86xx phones). Also, saved volume levels do not return to the default state after using the feature (86xx phones return to the default volume levels after use if you do not press the center of the Volume button to save the volume level).
- The 5212 and 5304 use half-duplex audio when using the speakerphone (both parties cannot talk at the same time). The 5224 and other 53xx phones use full-duplex audio.
- 52xx/53xx phones register input when buttons are released. 86xx phones register input when buttons are pressed.
- 52xx/53xx phones can display 20 characters on each display line—86xx phones display 16 characters.
- 52xx/53xx phones have a different set of ring tones than 86xx phones.
- 52xx/53xx phones do not have external speaker activation tones when feature buttons are pressed (for example, DND, Redial, Outgoing)—86xx phones do.
- 52xx/53xx phones have different feature error alert tones (for example, when an incorrect feature code is entered). 86xx phone alert tones resemble a “busy signal.” 52xx/53xx phones use four short tones for error alerts.
- The feature splash tone on 52xx/53xx phones (the tone heard on the handset when features are activated) interrupts the voice audio stream. With 86xx phones, the tone and the audio stream are heard at the same time (the tone is in the “background”). On a 52xx/53xx phone, feature activation tones supersede (interrupt) the audio stream.
- The DND alert tone generated on a 52xx/53xx phone differs from the DND alert tone generated on a 86xx phone.
- Speakerphone (handsfree) volume is generally lower on 52xx/53xx phones when compared to 86xx phones. This includes both DTMF and voice audio.
- The 5330 and 5340 phones have Navigation Page buttons located below the LCD to navigate between the three display pages. If you navigate to the second or third pages, you must use the Navigation Page buttons to return to the first page. (You are not automatically returned to the first page after using features or when the phone is idle.) The Home button (⌂) on the 5340 also returns you to the first page.

- The 5320, 5330, 5340, and 5360 phones have self-labeling buttons (they do not use designation cards).
- Several 52xx/53xx phone feature buttons behave differently than other model feature buttons. Mitel feature buttons descriptions are shown in the *MiVoice Office 250 Features and Programming Guide*.
- 52xx/53xx phones do not support the *Automatic* Network Address Translation (NAT) address type. In DB Programming, you must select either Native or NAT for private and public networks. Refer to the *MiVoice Office 250 Features and Programming Guide* to configure the NAT address type.
- On MiVoice Office 250 systems, the Mitel 5340 phone Backlight Timer is set for five minutes of inactivity (no button presses or audio, including music). After five minutes of inactivity, the display dims to 20% of the maximum brightness (the display backlight cannot be turned off). You cannot adjust these default settings.

See **Table 21** for a description of the 52xx/53xx phone feature buttons. Refer to the appropriate user guide for additional information on a specific model type.

Table 21: 52xx/53xx Phone Feature Buttons

Button	Action
(Up) (Down)	• Provide volume control for the ringer, handset, and speaker. • Scroll through options in some features.
(Speaker)	• Activates Handsfree Mode (speakerphone). • Activates features.
(Mute)	Temporarily turns off the microphone during a call.
(Applications/Directory)	**5320, 5330, 5340**, **and 5360**: Activates the Application menu. **5212/5312 and 5224/5324**: Activates the Directory.
(Special)	Activates features while on active calls. The (Special) button does not cancel features. The Star button (*) cancels features.
(Redial)	Calls the last *external* number dialed. Does not redial internal numbers.
(Hold)	• Places the current call on hold. • Left/backspace when entering dialpad characters.
(Transfer)	• Transfers the current call. • Right/forward when entering dialpad characters.
(Message)	• Connects to the voice mailbox and flashes when messages are waiting. • Toggles between Alpha Mode and Numeric Mode when entering dialpad characters.

Wireless IP Phones

The 8664, 8665, and 8668 are compact wireless IP phones. To operate, a NetLink SVP (SpectraLink® Voice check box Priority) server is required. For details, refer to the *SVP Server and Model 8664/8665/8668 Installation Manual*, part number 935.4521.

Mitel Cordless Devices

The MiVoice Office 250 supports the cordless handset and headset for the Mitel 5330, 5340, and 5360 IP phones. Both cordless devices connect to the Mitel IP phones through the cordless module, which attaches to the back of the phone. The cordless headset recharges in a headset cradle that attaches to the side of the phone. The cordless handset recharges in the handset cradle. For more information about the phone system features, refer to the *Mitel 5320/5330/5340 IP Phone User Guide,* part number 550.8123, and the *MItel 5360 IP Phone User Guide*, part number 550.8124.

NOTE Installation instructions are included with each device. However, when installing a cordless device, press and hold the Module Pairing button on the back of the cordless module for about 5-15 seconds. Then follow the “Hookswitch” instructions shown on the phone display to pair the cordless device with the module. See **Figure 5** for Module Pairing button and Hookswitch button locations.

Figure 5: Mitel Wireless Devices and Components

1 – Cordless module

2 – Cordless headset

5 – Cordless handset

6 – Cordless Module Pairing button

3 – Cordless headset charging cradle
4 – Cordless handset charging cover
7 – Cordless headset cradle cable

Support for 5610 Cordless Handset and IP DECT Stand

The Mitel 5610 Cordless Handset and IP DECT (Digital Enhanced Cordless Telecommunications) Stand provide MiVoice Office 250 basic SIP call processing features on a cordless handset. Although you can register up to eight handsets with an IP DECT Stand, the stand only supports up to three simultaneous calls. If three calls are already in progress via the IP DECT Stand, any other handset users who attempt to place a call will receive busy tone.

For site applications where all the users registered to a stand are expected to place calls frequently, register a maximum of three handsets to the stand.

For site applications where users seldom place calls, it may be suitable to register more than three handsets to a single stand. A ringing device is counted as a busy device.

Each handset is programmed as a separate extension (SIP phone) on the MiVoice Office 250, and they can be assigned to hunt groups or as Dynamic Extension Express associated destinations (like any other phone in the system).

The IP DECT stand connects to the network through the PC port on a Mitel 5312, 5320, 5324, 5330, 5340, or 5360 IP phone. To extend the wireless reception range, If needed, the IP DECT Stand can be equipped with an optional repeater.

For additional information and specifications, access the Mitel Online (MOL) Web site and navigate to the 5610 Handset and IP DECT Stand Web Page (http://tinyurl.com/ykky8px). Click the data sheet link at the bottom of the page. You will need your MOL user name and password.

Each handset associated with the IP DECT Stand requires a Category F license. The IP DECT Stand itself is not licensed. Also note that handsets are associated with a single IP DECT Stand, so they cannot be used across multiple IP DECT Stands.

Softphones

The MiVoice Office 250 supports the Unified Communicator® (UC) Advanced embedded softphone (on both Android and Apple iPhone devices), UC Express SIP Softphone, and 8602 softphone application.

Unified Communicator Advanced Embedded Softphone

Mitel UC Advanced integrates presence and availability, secure instant messaging, audio conferencing and web and video collaboration with the call control capabilities of your business communications system. The UC Advanced Desktop Client provides an embedded softphone that users can use with a USB headset or handset to place and receive calls.

For additional information and specifications, access the Mitel Online (MOL) Web site and navigate to the Unified Communicator Advanced Web page (http://tinyurl.com/2bk92xc). Click the data sheet link at the bottom of the page. You will need your MOL user name and password.

8602 SoftPhone Application

8602 application software is distributed separately from the system software. Performance and interoperability is equivalent to hard IP phones. For details, refer to the *Model 8602 User Guide*, part number 835.3019.

Digital Telephones

The following digital telephones can be installed on DEI equipment attached to the MiVoice Office 250:

- 8528 and 8568
- 8500, 8520, and 8560
- Basic, Standard, Associate, Executive, and Professional models

8528 and 8568

The newer Mitel 8528 and 8568 digital telephones replace the 8520 and 8560 digital telephones, respectively. The newer telephones function similar to the 8520 and 8560 with the following exceptions:

- New design look—see the images in Appendix D.
- Universal codecs (A-law and Mu-law interfaces) that do not require separate hardware revisions to support both U.S. and U.K. markets.
- Increased dynamic range and greater handset amplitude. Provides full compliance with FCC HAC and the American for Disabilities Act, as well as Section 255 of the Telecommunications Act and Section 508 of the Rehabilitation Act.
- Standard RJ11 Wiring Configuration—see page 112
- Downloadable firmware—see page 112
- Supports Mitel 12- and 48-Button Programmable Key Modules (PKMs)—see page 115

The 8528 and 8568 do ***not*** support the following peripherals or features that are applicable to the 8520 and 8560:

- PKM 50 (previously known as the 8450 50-Button Direct Station Select/Busy Lamp Field [DSS/BLF] unit)
- PKM 16 (previously known as the 8416 Mini-DSS unit)
- Personal Computer Data Port Module (PCDPM)
- Modem Data Port Module (MDPM)
- Off-Hook Voice Announce (OHVA)
- Dual-color LEDs. The System flag “Use Green LEDs for Direct Rings” does not apply.

The 8528 and 8568 are available in all English-speaking markets in which the 8520 and 8560 are available. These include English-speaking Canada, U.S., Australia, U.K., and European countries. See **page 371** for the telephone part numbers.

The 8528 and 8568 require v3.2 or later software and are not backwards compatible. As with the 8520 and 8560, the 8528 and 8568 do not require a license. And, like the 8520 and 8560, the 8528 and 8568 do not generate their own DTMF tones; instead they use available system DTMF transmitters.

Installation Instructions

An installation guide is packed with each Mitel 8528 and 8568 digital telephone. Refer to the instructions in the guide and to the information in this section. All Mitel installation guides are also available in PDF format on the Mitel eDocs Web site (http://edocs.mitel.com/).

Using a Headset

The 8528/8568 is designed to work with a headset that has a 4-pin, RJ-type connector. User satisfaction for any particular headset can be quite subjective, so Mitel recommends that users try before buying so they are more likely to be satisfied with the fit, quality, and cost of the solution. This recommendation includes all wired and wireless headsets. In addition, Mitel recommends using a headset with:

- a built-in microphone to avoid audio feedback or echo.
- FCC/IC hearing aid compatible (HAC).
- Certification Impact Analysis (CSA): UL 60950-1.

Wireless Considerations: A wireless headset must remain within the system's range operating limit from the base. Refer to the appropriate manufacturer's user guide for more details.

Cable Termination for the 8528/8568

Table 22 shows the cable termination for the 8528/8568. Unlike the 8520/8560, the 8528 and 8568 use standard RJ11 wiring configuration as defined in Telecommunications Industry Association (TIA) 968. The 8528 and 8568 are polarity insensitive. See the Installation Guide that accompanies each telephone for additional installation information.

Table 22: Cable Termination for the 8528/8568

RJ11C Pin	RJ11 Position at the Phone	RJ11 Position at the Wall Jack	Cable Color	Terminal Connector Color
1				
2				
3	Tip	Ring	Blue/White	Red
4	Ring	Tip	White/Blue	Green
5				
6				

Downloading Firmware

Unlike the 8520/8560 telephones, the newer 8528/8568 telephones use programmable firmware to update the phones. When a phone is powered up, the system checks the firmware file against the phone version. If the phone requires a firmware update due to a new firmware release being available or a system upgrade, the system downloads the updated firmware to the phone automatically.

There are two firmware images provided in the "cs-ep-mitl-85x8" firmware package. The image upgrades occur separately, and the phone is not usable during the process.

During the upgrade, the phone LEDs blink on and off, and the display shows a message to indicate which image is being upgraded and an estimate of how long the upgrade process will take from the time it started. The phone resets after each image upgrade is complete. The upgrade time varies depending on the number of phones undergoing the upgrade process at the same time. For example, on a heavily-loaded DEM, the entire process can take up to 13 minutes to complete. Do not unplug the phone until it has finished resetting and is fully operational.

If the firmware download process fails three times in a row, the system will stop trying to upgrade and simply allow the phone to come online. The system will then attempt to upgrade the phone the next time it resets (e.g., due to unplugging it or a system reset).

Specifications

For specifications, see the respective data sheets on Mitel Online (Products / Desktops / Digital Desktop Products) when they are posted and available.

Other Digital Telephones

Other digital telephones that can be installed on the system include:

- 8500, 8520, and 8560
- Basic, Standard, and Executive
- Associate and Professional

The *approximate* dimensions of the Standard Display, and Executive Display Phones are:

Height	3.0 in (7.6 cm)
Width	7.5 in (19.1 cm)
Length	9.0 in (22.9 cm)
Weight	2.3 lb (1.0 kg)

The *approximate* dimensions of the Associate Display, and Professional Display Phones are:

Height	3.25 in (8.6 cm)
Width	9.25 in (23.5 cm)
Length	7.75 in (19.7 cm)
Weight	2.09 lb (.95 kg)

The *approximate* dimensions of the Model 8662, 8622, 8620, 8660, 8560, and 8520 are:

Height	2.5 in (6.4 cm)
Width	8.8 in (22.3 cm)
Length	8.3 in (21.1 cm)
Weight	2.7 lb (1.2 kg)

The *approximate* dimensions of the Model 8500 and 8600 are:

Height	2.5 in (6.4 cm)
Width	6.4 in (16.3 cm)
Length	8.3 in (21.1 cm)
Weight	2.1 lb (0.9 kg)

Wireless Digital Telephones

The 8524, 8525, and 8526 are compact wireless digital telephones. For details, refer to the *8524/8525/8526 User Guide*, part number 935.4519.

Phone LED Flash Rates

Table 23, “Multi-Protocol Phone LED Flash Rates, IP Mode,” on **page 114** describes the LED flash rates for multi-protocol IP phones in IP mode in interruptions per minute (IPM). For more general information, refer to the *MiVoice Office 250 Features and Programming Guide*.

Table 23: Multi-Protocol Phone LED Flash Rates, IP Mode

	Steady 0 IPM	Slow 30 IPM	Medium 120 IPM	Fast 240 IPM	Flutter 60/960 IPM
CALL		Call is active	Call is recalling from hold or transfer	Call or queue callback is ringing in or camped on (waiting)	Call is on hold at the user’s phone or the user has entered a transfer feature code.[1]
CNF		Conference is in progress.	Conference is recalling your phone.		A conference party is on conference wait hold.
DATA[2]	Auto-dial modem is dialing and the data port is unavailable. (Red LED)			Ring signal is being sent to the data device.	Data call is in progress and accessible. A continuous flutter indicates that the data device is off-hook.
DND	Phone is in Do-Not-Disturb.				
PKM 16 (MINI-DSS)[2]	The programmed phone in busy or unplugged.	The programmed phone is in DND.		The programmed phone has a call ringing in.	Continuous flutter indicates that the programmed phone was left off-hook.
FEATURE[2]	The feature is in use.				
FWD	Calls are being forwarded.				
HUNT GROUP	All phones in hunt group are unavailable.			Hunt group has a call camped on.	
IC		Intercom call is in progress.	Recalling intercom call is camped on (waiting).	Intercom call is camped on (waiting).	An intercom call is on hold at the user’s phone or the user has entered a transfer feature code.[1]

(Sheet 1 of 2)

Table 23: Multi-Protocol Phone LED Flash Rates, IP Mode (Continued)

	Steady 0 IPM	Slow 30 IPM	Medium 120 IPM	Fast 240 IPM	Flutter 60/960 IPM
MSG	Phone is in alpha-numeric mode.		Message is waiting.		
MUTE	Microphone is muted.				
SPEAKER	Speakerphone is on or ready for use. Or, phone programming is in progress.				
SPEED DIAL[2]	Designated phone is busy or unplugged.	Designated phone is in DND.		Designated phone has a call ringing in.	Continuous flutter indicates that the designated phone was left off hook.
TRUNK	Trunk is in use at another phone, in a conference at your phone, or it is unplugged.	Trunk is in use at your phone.	Trunk is recalling from hold or transfer.	Trunk is ringing in, camped on (waiting), or in busy trunk callback queue to your phone.	Trunk is on hold.

(Sheet 2 of 2)

1. For transfers, the IC or CALL button LED goes out when the user hangs up.
2. Requires a programmable button.

Programmable Key Modules

The Programmable Key Modules (PKM) are peripheral devices that provide additional programmable buttons for the 5224/5324 IP phones and the 8528/8568 digital telephones. There is a 12-button PKM and a 48-button PKM. The PKMs connect to the phones using the modular cable included with the PKMs and an IP PKM Interface Module (IM) installed in the phones.

With the firmware included in the v5.0 software release, the 5330 and 5340 IP phones now also support the 12-button and 48-button PKMs.

A total of two PKM 48s may be attached to these phones, for up to 96 additional buttons. You cannot connect a 12-button PKM to a 48-button PKM and you cannot connect two PKM 12 units together. Supported configurations are as follows:

- one PKM 12
- one PKM 48
- up to two PKM 48

Each PKM button may be programmed as a feature button, a Direct Station Select (DSS) button, a Speed Dial button, or for other uses. Each button has a status indicator.

Figure 6 shows examples of the PKM 12 and PKM 48.

Figure 6: PKM 12 and PKM 48

PKM 12

PKM 48

Specifications

For information on the Mitel PKM 12 and 48 specifications, access the Mitel Online Web site and navigate to the Mitel 12 and 48 PKM Web page (http://tinyurl.com/cae7ka). Click the data sheet link at the bottom of the page. You will need your MOL logon user name and password.

Direct Station Select/Busy Lamp Field Units

The MiVoice Office 250 supports two 60-button DSS/BLF Units and one 50-button PKM (previously known as the 50-button DSS/BLF Unit). Although you can use either type of unit on any of the older digital telephones, the type of phone determines which is the more appropriate unit to use.

- **With 8520 and 8560 digital telephones**: PC Data Port Module (PCDPM)-equipped 8520 and 8560 phones support up to four PKM 50 units (previously known as the 8450 50-button DSS/BLF Unit), part number 550.8450. Up to 20 of the 5-rows-by-10-columns PKM units can be installed on a system. Each unit is powered by a 450 mA AC transformer and is connected to a PCDPM installed in the phone. For PCDPM installation instructions, see “Installing a PCDPM” on **page 1-346**.
- **With Executive, Standard, Professional, and Associate Display digital telephones**: PCD-PM-equipped Executive, Professional, Standard, and Associate display phones support up to four 60-button DSS/BLF units each. Up to twenty 60-key DSS/BLF Units can be installed on a system, or node. Each unit is powered by a 450 mA AC transformer and connected to the PCDPM

installed in the phone. The 6-rows-by-10-columns model, part number 550.4200, is designed for Executive and Standard Display phones. The 12-rows-by-5-columns model, part number 560.4000, is designed for Professional and Associate Display phones.

PKM 50 and 60-button DSS/BLF units always require dedicated AC transformers. For a diagram of AC transformer requirements, see Figure 8 on **page 348**. For installation instructions, see "Connecting DSS/BLF Units" on **page 1-357**. Unit dimensions appear in **Table 24**.

Table 24: 60-Button DSS/BLF and PKM 50 Dimensions

	60-Button DSS/BLF	PKM 50
Height	3.0 in. (7.6 cm)	2.5 in. (6.4 cm)
Width	7.5 in. (19.1 cm)	6.4 in. (16.3 cm)
Length	9.0 in. (22.9 cm)	8.3 in. (21.2 cm)
Weight	1.7 lb (0.8 kg)	1.2 lb (0.6 kg)

Each unit provides one-button access to 50 or 60 extension numbers, trunks, features, or other resources, and four units provide access to 200 or 240 extensions, features, and resources. Numbers programmed for the buttons can be phone extension numbers, hunt group numbers, trunk access codes, or feature codes.

The layout of the buttons and the numbers are assigned using one of 20 keymaps in DB Programming. The intercom number/button assignments of each keymap are programmed in four lists, pages 1–4. For example, single units are assigned page 1 of the desired keymap, tandem units are assigned two pages of the desired keymap—page 1 for the first unit and page 2 for the second unit, and so on. For programming instructions, refer to the *MiVoice Office 250 Features and Programming Guide*.

Together, the lamps in the buttons create a Busy Lamp Field that indicate the status of each station, hunt group, or other feature/resource assigned to the buttons. If assigned to a station, the LED in the button is solidly lit when the associated station is busy or unplugged, flashes slowly when the station is in Do-Not-Disturb (DND), flashes rapidly when the station has a call ringing in, or flutters continuously if the station is causing a STATION OFF-HOOK system alarm.

If assigned to a hunt group, the LED indicator is solidly lit when all stations in the hunt group are unavailable, either busy, in DND, or removed from the hunt group. The LED indicator flashes rapidly when a call is camped-on to a hunt group. If assigned to a feature or resource, the LED indicator is lit to show busy status and unlit to show idle status.

NOTE Group Call Pick-Up does not affect DSS lamp status any differently than standard hunt group calls or individual calls.

8416 Mini-Direct Station Select (Mini-DSS) Unit

The MiVoice Office 250 supports a PKM 16 (previously known as the 8416 Mini-DSS Unit) that provides one-button access to 16 extension numbers and/or feature buttons/resources. The optional PKM 16 can be attached to 8520, 8560, and 8660 phones.

For reference, the following information matrix shows the *approximate* dimensions of the unit.

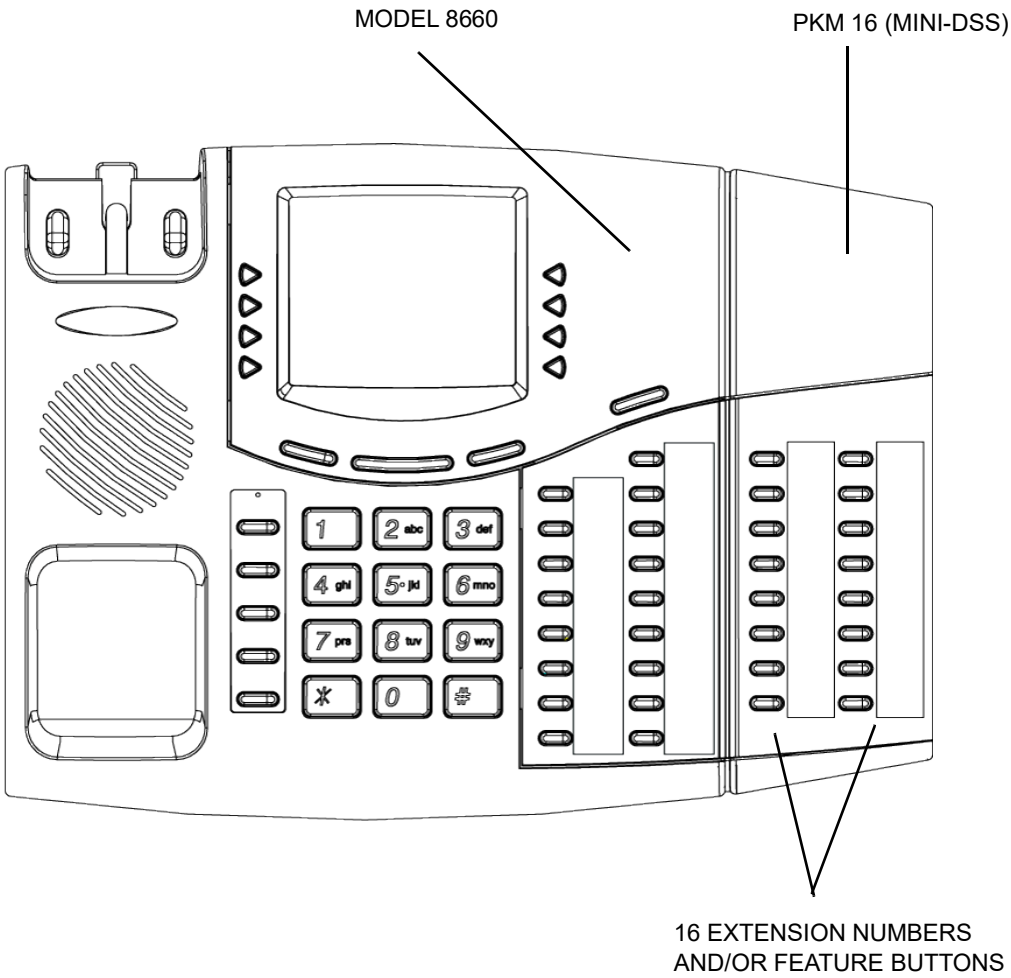
Table 25: PKM 16 (8416 Mini-DSS Unit) Dimensions

Dimension	Measurement (Metric)
Height	1.5 in. (3.8 cm)

Table 25: PKM 16 (8416 Mini-DSS Unit) Dimensions (Continued)

Dimension	Measurement (Metric)
Width	2.6 in. (6.5 cm)
Length	7.7 in. (21.1 cm)
Weight	2.1 lb (0.13 kg)

Figure 7: PKM 16 (8416 Mini-DSS Unit)



5310 IP Conference Unit

The optional 5310 IP Conference Unit can be connected to the 5224, 5324, 5330, 5340, and 5360 IP phones to provide full-duplex, advanced IP conferencing (see **Figure 8** below for an example, and see page 371 for part numbers).

For additional information and specifications, access the Mitel Online (MOL) Web site and navigate to the 5310 IP Conference Unit Web page (http://tinyurl.com/ybswtof). Click the data sheet link at the bottom of the page. You will need your MOL user name and password.

For the 53xx IP phones, the conferencing unit is comprised of a saucer and a module that connects to the back of the phone. For the 5324 IP phone, an optional remote control mouse is also needed

for full feature functionality. For the 5224 IP phone, the conferencing unit is comprised of a saucer and an optional side control unit that attaches to the right side of the phone.

NOTE Although pressing the fixed Mute button on the phone itself mutes the audio on the attached 5310 conferencing unit, the orange mute LED indicator on the conference unit does not light up. However, if using the Conference Controller application (5330, 5340, and 5360), the optional remote control mouse (5324), or the side control unit (5224) to mute the conference unit, the orange mute indicator does illuminate as designed.

Figure 8: Mitel 5310 IP Conference Unit

Line Interface Module (LIM)

The LIM (see table below) is an optional device that adds analog trunk capability to the Mitel 5224, 5324, 5330, 5340, and 5360 IP phones (see **Figure 9** below for an example). In the event of a network failure, the phone can be configured to use the public switched telephone network (PSTN) to provide continued basic telephony until service in restored. The analog line can also be used for emergency calls and local calling.

LIM Part Number	Markets
50004197	Canada, Caribbean, North America, Netherlands, France, Australia, Luxembourg, Finland, Denmark, Austria
50004198	UK, Ireland, Italy, Belgium, Germany, New Zeland, Portugal, Spain, Norway, Sweden

The LIM can be used to:

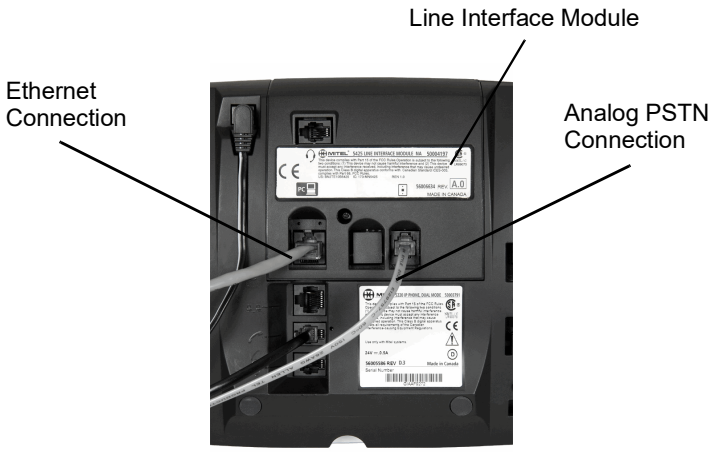
- allow incoming and outgoing analog PSTN calls directly to and from the IP phone
- provide survivability (failover) in the event IP connectivity is lost
- provide emergency dialing support (emergency calls can be programmed to connect through the analog PSTN connection)

Each IP phone equipped with a LIM can be programmed with one of the following emergency call routing preferences (refer to the *MiVoice Office 250 Features and Programming Guide* for programming details).

- **System Only:** Emergency calls are routed using the system's designated emergency call routing rather than using the analog trunk attached to the LIM. This is the default emergency call handling rule for all phones that do not have a LIM attached.
- **LIM First:** Emergency calls are first routed through the LIM. If unsuccessful, calls will then be routed through the system.
- **LIM Only:** Emergency calls are routed through the LIM only. If unsuccessful, the display indicates that the call failed.

NOTE When using the analog line attached to the LIM, the phone's functionality is reduced to that of a basic telephone without any features. For example, users are unable to change volume, mute calls, transfer calls, put calls on hold, etc.

Figure 9: Line Interface Module



IP Single Line Adapter (IP SLA)

This product is no longer available from Mitel. Please login to the Mitel Solutions Alliance (MSA) site through Mitel Online, and review the SIP certification documentation available through the MSA Downloads link.

Optional Handset Amplifiers

The typical handset amplifier is an external device that plugs into the single line phone where the handset is normally connected. The handset is then plugged into the amplifier. Receiver voice volume is controlled by turning a thumbwheel or similar control located on the amplifier. Such amplifiers are generally equipped with a transformer that requires a 110 VAC outlet. For installation instructions, see "Installing a Single Line Handset Amplifier" on page 367.

The 800-ohm loop limitation is usually suitable for accessing on-premises phones from off-premises phones. However, for applications where one off-premises phone will be calling another through the T1 span, the impedance between the two off-premises phones (both circuits added together) must be less than 800 ohms (loop measurement). In this situation, a Class A or B FIC circuit is preferable.

When an IP SLA circuit is designated for OPX use in DB Programming, its voice volume levels are increased approximately 3 dB over normal single line phone voice volume levels. This increase in

volume level attempts to compensate for any loss in the external OPX circuit. Refer to the *MiVoice Office 250 Features and Programming Guide*

However, in certain off-premises applications, the voice volume levels may still be unacceptable. The degradation in voice volume is due to the natural voice frequency range limitations of the telephone company or customer-provided circuit. To increase the frequency range, Mitel recommends installing a two-wire, negative impedance repeater unit. Such units typically provide from 0–15 dB of voice volume gain and allow regulation of the gain in each direction when simultaneous voice transmission occurs. The switching sensitivity on the unit should also be adjustable.

The repeater unit is installed at the MDF between the station block and the cabling to the OPX phone. For OPX installation instructions, see “Connecting CO Trunk Cables to the MDF” on page 153.

When ordering a repeater unit, consult with the supplier for ordering the proper mounting shelf and power supply for the unit. Typically, the product includes additional information about operating and adjusting the repeater unit.

NOTE Each OPX circuit can have extended ring cadence enabled or disabled. For details, refer to the *MiVoice Office 250 Features and Programming Guide*. The extended setting lengthens the duration of the ring signal to meet the requirements of many OPX and/or repeater applications. The default setting is ***disabled***.

Playback Devices

AC-ringing playback devices may be connected to the single line port on the chassis or to IP SLAs. Each IP SLA supports one device. A playback device is an answering machine that answers the call, plays a message, and then disconnects from the call. Playback devices are installed like single line phones, using one-pair twisted cable and four-conductor modular jacks. The device must be able to detect ringing, provide ring trip, and automatically disconnect when the announcement is completed. For IP SLA installation instructions, see “Installing an IP SLA” on page 304. For installing a selected playback device, refer to the manufacturer’s instructions.

NOTE For Automated Attendant features and for optimal playback device performance and functionality, use an optional Unified Voice Messaging (UVM) or NuPoint Unified Messaging (UM) system and applications software instead of IP SLA circuits.

Playback devices may be used with hunt groups to speed up call processing. Hunt groups can have three types of special phones: announcement stations, overflow stations, and recall destination stations. These types of phones may be equipped with a playback device that answers the call, then disconnects to transfer it back to the hunt group. For more information, refer to the *MiVoice Office 250 Features and Programming Guide*.

Playback devices are programmed as regular single line phones that are assigned an extension number. If the device is sensitive to Camp On tones and disconnects calls, the programmer can disable the Camp On for that phone circuit. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.

Attendant Console

With the MiVoice Office 250, the attendant function can be served by a software application operating on a personal computer (PC) or by a multiline display phone and attachment.

Personal Computer Application

The Attendant Console is a special software program and hardware security key that allows attendants to use a PC to answer and manage incoming call traffic. Although designed for use with larger communication systems, the Attendant Console provides enhanced functionality such as electronic text messaging and directory dialing for smaller systems, as well.

The Attendant Console PC, which requires a System Level 2 OAI interface, is connected to the system through a TCP/IP connection. For details about equipment functions and operation, refer to the *Attendant Console User Guide* supplied with the console software.

Attendant Service Phone

To achieve Attendant-like capability on the MiVoice Office 250, Mitel recommends using a six-line display phone with an attached PKM (DSS Unit). The phone is equipped with a jack for plugging in a Mitel handset or headset.

To operate correctly, the phone ***must*** be programmed for headset mode. This can be done in one of the following ways:

- Enable Headset Mode in Phone Programming in the system database.
- At the phone, unplug the headset or handset to take the phone off-hook and enter the Headset On/Off feature code (317) to enable headset mode. Then, plug in the headset or handset.

NOTE Using the Headset On/Off feature code at the station, the user can disable the headset mode.

If the user wishes to receive audible signals at the headset, a programmable phone flag can be set to send a single tone to the headset when a call rings in. For details, see the phone programming flags in the *MiVoice Office 250 Features and Programming Guide*.

IP Network Requirements

This section provides a brief technical overview of data network characteristics that will help to minimize faulty operation of the MiVoice Office 250. For additional technical details and applicable procedures designed to promote an adequate IP network environment, see **Private Networking** and **Network IP Topology** appendices in this manual.

Network Capability Analysis

Because the Mitel IP devices use IP to connect to other IP devices, the data network must meet certain specifications that support an adequate VoIP network. This section summarizes adequate network requirements.

NOTICES

The following notices apply to the network capability:

- Network security is the responsibility of the customer's network administrator. Mitel is not responsible for network problems due to security violations involving the IP address of the Mitel IP devices. This includes, but is not limited to, toll fraud and interrupted network service.
- Due to the volatility in delay, jitter, and packet loss, sending VoIP traffic over the Internet may result in intermittent poor voice quality, dropped calls, and equipment resets. No delivery guarantee can be made for VoIP traffic over the Internet.

Run your preferred Network Monitoring software to monitor and analyze network traffic for at least 24 hours to assess the network prior to purchasing or installing the hardware required to set up VoIP.

Performance Criteria

The acceptable quality of an IP telephone call depends on the adequacy of the data network through which it travels. Network adequacy for VoIP transmission includes the speed and capacity of the network configuration, type of connection, routing scheme, and amount of network traffic.

Mitel requires that its business-based VoIP products and solutions be used only on data networks that meet the near toll quality and business communication quality criteria outlined in this section. Successful VoIP platform deployment requires installation in a correctly provisioned, configured, and tuned data network.

For both near toll quality and business communication quality levels, the available network bandwidth/capacity should provide approximately 32 kbps per call using the G.729 vocoder and approximately 86 kbps using the G.711 vocoder. These capacity values are based on a 30 ms packetization interval.

Mitel recommends the following network parameters as measurable performance criteria for an adequate VoIP network:

- **Network Packet Loss**: The percentage of uniformly distributed data packet loss (nonburst). For near toll quality service, uniformly distributed data packet loss should not exceed 1 percent. For business communication quality service, uniformly distributed data packet loss should not exceed 3 percent.
- **Network Delay**: The data network delay between two peer-to-peer network devices. For near toll quality service, one-way delay (UDP–User Datagram Protocol–traffic) should not exceed 80 ms,

assuming all jitter buffers are set to accommodate 40 ms maximum jitter level. For business communication quality service, one-way delay (UDP traffic) should not exceed 120 ms, assuming all jitter buffers are set to accommodate 80 ms maximum jitter level.

- **Network Jitter**: Variance in delay. For near toll quality service, average jitter should not exceed 20 ms and maximum jitter should not exceed 40 ms. For business communication quality service, average jitter should not exceed 40 ms and maximum jitter should not exceed 80 ms.
- **Hops**: The number of router points a data packet must pass through during transmission. Hops should not exceed 15 in order to minimize the number of potential problem points.
- **Committed Information Rate (CIR) and Burst Rate (BR)**: 512 Kbps/each. This criterion means that 512 Kbps will always be available, and the link may achieve a BR of up to 1024 Kbps for short periods if the bandwidth is available from the WAN provider. Typically, there is no guarantee of BR availability. Voice traffic should stay below the CIR value. When voice traffic exceeds the CIR value, packets may begin to queue within the Frame Relay network and become subject to delay. These packets could eventually be tagged as Discard Eligible (DE). Packets marked as DE could be discarded before reaching their destination. The potential queuing delay and packet loss can adversely affect voice quality.

Mitel also recommends the following PSTN performance characteristics:

- **Loop Current**: 18–20 mA
- **Line Voltage**: 42.5–56.5 V
- **Ring Voltage**: 55–130 Vrms (Class A)
- **db Level**: Must pass at least +3dBm without distortion into a 600-ohm load

All of the preceding PSTN requirements are consistent with the EIA/TIA-464B specifications. Use Ethernet switching mechanisms rather than Ethernet hubs. Switches offer inherent Ethernet advantages over hub (shared) technology. Switches provide advantages such as segmentation, full-duplex operations, Layer 2 prioritization, etc.

IMPORTANT	Using background music and paging on a hard IP phone, such as the IP PhonePlus, consumes as much bandwidth as a telephone call to the device. If the site has limited bandwidth, make sure IP phone users do not enable background music and that they limit using the paging feature.

For any router that will have VoIP traffic passing through it, program the router to prioritize the packets sent by the IP devices. In addition, program the external router to fragment large data packets into smaller packets. This reduces the possibility that voice packets will have to wait for a large data packet at the router.

Automatic NAT Detection

Network Address Translation (NAT) is a method for translating internal IP addresses or network addresses into public Internet addresses.

The Automatic Network Address Translation (NAT) Detection feature allows 52xx/53xx phones to operate inside or outside a private network NAT or firewall without having to change the IP Settings\ NAT Address Type field every time the phone is relocated. The feature enables Mitel hard IP phones and the Mitel 8602 IP softphone to place and receive calls either from inside or outside a Mitel private network.

Mitel 52xx/53xx IP phones do not support the *Automatic* Network Address Translation (NAT) address type. In DB Programming, you must select either Native or NAT for private and public networks. Refer to the *MiVoice Office 250 Features and Programming Guide*, to configure the NAT address type for Mitel 52xx/53xx IP phones.

Automatic NAT Detection automatically determines whether an IP phone is inside the firewall or outside the firewall. In DB Programming, the administrator designates each IP phone to use the Auto type, as well as either Native or NAT types. The default selection is Auto. Automatic NAT detection is performed each time an IP phone comes online and whenever the feature is set through DB Programming.

This feature does not work with IP Gateways, IP trunks, wireless 8664/8665/8668 phones, any IP phones in SIP mode, and third-party SIP phones. Firmware on these phones cannot run the Auto NAT Detection test. For these phones, the installer needs to manually select the NAT Type setting.

Protocols and Ports

Port Usage and Protocols on page 442 in Appendix B describes required connections for IP phones, and UDP port assignments. The IP network must allow the phones to establish these connections for proper operation.

99 Nodes Support

Version 2.1 or later software allows MiVoice Office 250 systems to expand to 99-node networks in the United States and European markets. Each node can transparently interconnect to 98 other systems using T1/E1/PRI networking and IP networking.

> **NOTE** Due to licensing changes in v4.0 or later, the Extended Node Capacity license is no longer required to support more than 63 nodes.

With the 99 Nodes feature, the increased network capacity does not affect phone user features. Phone users can still dial an extension of a remote user, and the system will transparently route the call throughout the network. From an administrative perspective, you must consider the following issues as the node capacity increases:

- **Database Management**: To program the 99 Nodes feature, you must log-in to individual systems to configure them. You must plan for supporting a 99-node network by devising a numbering plan scheme for each node. You should be aware that the creation of new devices automatically causes the system to export extensions to the network. The use of export/import may take longer if you attempt to export/import with all 98 remote nodes.
- **Numbering Plan**: The system supports extensions with a maximum of five digits. It is important to design a network numbering plan that can support all devices in the network.
- **DB Programming Export/Import**: Creation of new devices on any node of a network automatically causes the system to export extensions across the network. Once network established, you can use DB Programming to adjust node parameters, including the import or export of devices. Because of the potential complexity of managing a network with numerous nodes and devices, it is essential that the system administrator devise a network numbering scheme that accommodates all 99 nodes.

- **Networking Resources (IP and PRI)**: You must be aware of the Primary Rate Interface (PRI) and Voice over IP (VoIP) resources required to maintain a 99-node network. For VoIP resources, all nodes must have the appropriate IP bandwidth to support the maximum IP networking calls.
  - *T1/E1/PRI Networking*: Networking across T1/E1/PRI remains the same as in previous versions, except for the increased node capacity. With appropriate licensing, in the T1/E1/PRI field in DB Programming, you can number network nodes from 1 to 99.
  - *IP Networking*: IP Networking functionality remains the same as in previous versions, except for the increased node capacity. If a device does not have an IP resource reserved, the system dynamically allocates one from the pool of available IP resources.
- **OAI Applications**: To support more than 63 nodes, an appropriate version of the CT Gateway Server is required. Performance degradation may occur on the CT Gateway and individual nodes depending on the number of OAI applications that connect to the CT Gateway. If many OAI applications are connected directly to an individual node, slowdowns on those nodes may occur.

IMPORTANT	The CT Gateway Server must meet Mitel performance requirements when it is attached to a 99-node network. When configuring a 99-node network, this could increase the OAI data by 50%. For example, the Attendant Console application requires a large amount of data at power-up. Multiple Attendant Console applications, along with fully populated nodes, could increase OAI data by 50% during the Attendant Console power-up sequence.

Component Compatibility for 99 Nodes Support

See the latest *MiVoice Office 250 Features and Programming Guide* for a table that lists component version numbers that have been validated to support the 99 Nodes system feature.

Peripheral Equipment

The MiVoice Office 250 supports peripheral, or external, equipment that provide enhancements to the IP system's basic business communication services. The following sections describe equipment that can be attached to the platform.

Voice Processor

NOTICE

The Voice Processing Unit (VPU): The VPU is no longer supported in v3.0 and later software. The VPU was discontinued in May 2007. Mitel recommends that current VPU installations upgrade to Unified Voice Messaging (UVM) or NuPoint Unified Message™ (UM). Contact your local provider for more information.

NuPoint UM is voice mail software that resides on the Mitel Applications Suite® (MAS) server. MAS is a software solution that provides co-residency features for applications that use the Mitel Standard Linux® (MSL) operating system. Refer to the latest *Mitel Applications Suite Installation and Maintenance Manual* for software specifications and detailed installation information. Refer to the *MiVoice Office 250 and NuPoint UM Integration Guide*, part number 580.8008, for information on setting up NuPoint UM as the voice mail system for the MiVoice Office 250. The following MiVoice Office 250 voice mail applications and features are not supported in v3.x with NuPoint Messenger:

- Automatic Speech Recognition (ASR). Refer to the Mitel Speech Auto Attendant (SAA) product, provided with NuPoint Messenger for similar functionality.
- E-Mail Reader
- Soft menu buttons on the phones when making internal calls to voice mail
- MiVoice Office 250 v3.2, in combination with MAS 2.0, provides support for the NuPoint UM Record-a-Call (RAC) feature and includes phone display and soft menu button support when making calls to the NuPoint UM voice mail.

SMDR/Error Message Output

Output data from SMDR and Error Message Recording features is accessed through an IP socket. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

CT Gateway

System OAI Level 2 applications can be used in conjunction with the CT Gateway. The CT Gateway provides a TCP/IP System OAI interface that supports a network. As far as the System OAI streams are concerned, each node in a network functions individually. The user cannot program a System OAI connection to Node 1 that will control devices or receive events for other nodes. If the user needs to control devices on multiple nodes, they must install a CT Gateway. The CT Gateway receives the OAI streams from multiple nodes and consolidates them into a single stream for a System OAI application. The CT Gateway sends and receives all of its data via TCP/IP connections.

For details, refer to the *CT Gateway Installation Manual,* part number 835.2161. For environmental requirements, see **page 60**.

Additional Information

If necessary, contact the Mitel DataNet CommSource division for additional information about the following optional equipment:

- Standby Power (system battery back-up)
- Uninterruptible Power Supply (UPS)
- Lightning and AC Surge Protection
- Headsets
- Paging Equipment
- Single Line Phones
- Cordless Telephones
- Music-On-Hold
- OPX Repeaters
- Facsimile Machine
- Playback Devices

Chapter 5

Installation

Chapter Overview

This chapter provides information on installing, upgrading, and expanding the MiVoice Office 250 and its various parts, including the expansion modules, IP, digital, and analog single line phones, and optional system components. For system specifications and a description of the overall system, see the **Specifications** and **Product Description** chapters, respectively.

The chapter is organized according to the types of tasks required to install, upgrade, expand, and maintain the MiVoice Office 250. Tasks are grouped into the following categories:

- Installation Planning on **page 132**
- MiVoice Office 250 Base Server Chassis on **page 137**
- Base Server Modules on **page 180**
- Installing the Processing Server (PS-1) on **page 201**
- Digital Expansion Interface (DEI) on **page 209**
- Station Cabling Connections on **page 233**
- Maintenance Tasks on **page 247**
- Phones on **page 257**
- System Upgrades and Expansion on **page 307**
- Optional System Equipment on **page 328**

Please read all information leading up to the procedures, paying close attention to any included cautions and notes. For information about system requirements and capacities, see the **Specifications** chapter. Figure 14 on **page 144** provides an illustration of the unit chassis.

Technical Certification Requirement

As a Mitel 5000-certified technician, you should be familiar with MiVoice Office 250 platform architecture and operation. And, you should be able to use the Web-based technical support features provided by Mitel. For example, the installation and upgrade procedures require you to navigate the following Web sites:

- Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome)
- Application Management Center (http://www.ebiz.mitel.com/amclogin.jsp).
- Knowledge Base Center (accessible through MitelOnline)

NOTICE

This manual is designed for Mitel certified field technicians. It does ***not*** provide detailed instructions for carrying out tasks typically performed by competent telephone technicians. For example, the manual does not include premises wiring practices as required by the National Electrical Code [BS 6701 in Europe]. Excluded from this manual are procedures for cable layout and installation, AC power installation and appropriate grounding, how to eliminate or prevent the effects of RFI, EMI, lightning, AC power disturbances or static discharge, and other telephony practices considered standard within the industry. Cable installers, electricians, and field technicians are expected to be properly trained and, if applicable, licensed in their trade practices. *Mitel system certification is required of any person requesting Mitel technical assistance.*

INSTALLATION PLANNING

This section provides procedures for planning the installation of an HX Controller or CS Controller platform. If upgrading or expanding an existing system, see “System Upgrades and Expansion“ on **page 307**.

Network Installation Concerns

To install a network, you will need to acquire the following components:

- **Software License**: Software features, including networking, are controlled by a software license. The license uploaded to the system must have the appropriate networking feature.
- **T1 or E1 Service**: For each network span, order a clear channel, point-to-point T1 or E1/PRI span from the service provider (not PRI service, even though you will be using T1M or T1M-2 modules set up for T1 or E1/PRI). If you intend to configure the T1/E1 module for PRI use, you must have the PRI software feature license, part number 840.0227.
- **T1M or T1M-2 Modules**: For a private network installation, each network span requires two T1M or T1M-2 modules, part number 580.2700 or part number 580.2702: one for each end of the span.
- **Processor Module**: The HX Processor Module (PM) or the CS Processor Module (PM-1) are removable units. The Processor Module is incorporated into a removable main board but can be removed from the chassis, and the PM-1 is a separate card that is added into the CS base chassis for it to function. If necessary, a Processor Expansion Card (PEC-1) can be installed on top of the HX PM or CS PM-1. For details, see the HX Controller procedure for “Adding a Processor Expansion Card (PEC-1)” on **page 1-181** or the CS Controller procedure for “Adding a Processor Expansion Card (PEC-1)” on **page 1-186**.

You can create a private network through IP resource programming—Refer to the *MiVoice Office 250 Features and Programming Guide*. Installation recommendations include the following:

- Each node ***must*** have at least one central office (CO) [local exchange] trunk. Emergency calls must be programmed to use the local trunk. See page 337 for additional Emergency call information. Also, refer to the *MiVoice Office 250 Features and Programming Guide* for information on configuring SIP Peer trunk groups for emergency calls.
- All nodes should have access to the node where an optional external voice processing system is located, with a minimum number of hops through other nodes. This will have a significant impact on the network layout. The network does not allow one node to use resources from another node (for example DSP resources) with the exception of external page ports.

NOTICE

With regard to IP Addresses, subnet mask, and gateways, it is not necessary that the CT Gateway be on the same LAN as the MiVoice Office 250 system IP address and subnet mask.

Pre-Installation Checklist

Use the following checklist when preparing to install the system. System specifications, including capacities are included in the **Specifications** chapter. System descriptions appear in the **Product Description** chapter.

Establishing Suitable Environmental Conditions for the System

Follow these guidelines to establish suitable environmental conditions for the system:

- Locate the chassis within 5 ft (1.5 m) of an isolated, dedicated, 105–125 VAC, 57–63 Hz, 15 A, single-phase commercial power source. Also, use an uninterruptible power supply (UPS) with the Mitel 5000 chassis.

WARNING

Electric Shock Hazard

- Never install telephone wiring during a lightning storm.
- Never install telephone jacks in wet locations unless the jack is specifically designed for wet locations.
- Never touch uninsulated telephone wires or terminals unless the telephone line has been disconnected at the network interface.
- Use caution when installing or modifying telephone lines.

Not following these suggestions could result in serious personal injury.

NOTICE

The system must be connected to an isolated, dedicated AC circuit. Power, neutral, and ground wires must run separately from the outlet to the breaker panel without being bonded to any other wire or circuit.

- Do ***not*** plug any other equipment into this outlet. To maintain the protection provided by the isolated, dedicated circuit, the AC power cord length limits the distance between the power supply and the outlet.
- Do ***not*** use an extension cord.

Do ***not*** defeat (or leave floating) the ground wire in the AC power cord. To protect the system from AC voltage surges, install a surge/spike protector having the specifications cited on page 62.

- Select the chassis location to minimize cable run length.
- Do not expose the chassis to direct sunlight, high humidity, heat, dust, or strong magnetic fields (such as those generated by heavy motors and large copy machines).
- The MDF requires a ¾-inch (2 cm) plywood backboard for mounting system equipment, peripherals, and distribution blocks. Usually, a board measuring 4 ft × 6 ft (1.2 m × 1.8 m) provides sufficient space. However, **a larger backboard may be required for mounting multip**le AC transformers, IP SLAs, or **other system-related devices.**
- Provide ample air space around the chassis. Do not place or store items on top of the chassis.
- Install a UPS with the CS controller.
- Locate the equipment in a climate-controlled room. Mitel recommends that the maximum operating temperature (as stated below) *never* be exceeded. Therefore, when installing the chassis

and any other system equipment, allow a sufficient margin for error (for example, air conditioning failure, routine maintenance, plant shutdown, etc.) As a general rule, if conditions are suitable for office personnel, they are also suitable for all equipment. A properly controlled environment will help to extend the equipment operating life.

Table 26: Equipment Environmental Requirements

Requirements	In Operation	In Storage
Temperature		
Equipment Chassis	32 to 104 °F (0 to 40 °C)	+40 to 185 °F (–40 to 85 °C)
Station Instruments	32 to 113 °F (0 to 45 °C)	–40 to 185 °F (–40 to 85 °C)
Relative Humidity (Non-Condensing)	5–95%	5–95%
Altitude	Up to 10,000 ft (3048 m)	Up to 40,000 ft (12,192 m)

Gathering the Necessary Tools and Supplies

Keep the following information in mind when gathering the necessary tools and supplies to install a system:

- A high-impedance digital multimeter to ensure correct wiring of the modular jack assemblies [line jack unit]. Multimeter accuracy must be ±0.5% or better.
- With a U.S. system, standard terminal blocks (66-type) and bridging clips for constructing the MDF. With a European system, Krone blocks for the MDF jumper field.
- With a U.S. system, industry-standard, 2-pair non-stranded cable terminated with modular plugs for connecting the MDF terminal blocks to single line jacks on the back of the CS controller.
- Industry-standard cross-connect, or jumper, wire may be needed for certain applications.
- Gas discharge tubes with silicon avalanche suppressors for lightning protection on the CO [Local Exchange], DID (USA only), and OPX trunks.
- Grounding terminal and 10AWG [diameter 6mm] wire for grounding the chassis.
- Standard hand tools and the mounting hardware for the MDF backboard, trunk and termination blocks, modular jack assemblies [line jack unit], etc.
- For online or offline system programming, a computer with the appropriate specifications (see **page 127)**.
- Industry-standard, one-pair (two-conductor) twisted cable to run from the MDF to all station instruments and four-conductor modular jack [line jack unit] assemblies for all digital and analog station instruments, including digital telephones, single line DTMF phones, SLAs, and playback devices.
- Grounding terminal and 10AWG (diameter 6mm) wire for grounding the chassis, Single-Line Adapters, etc.

If installing any T1 or E1 and/or Primary Rate Interface (PRI) trunks: Depending on the type of T1/PRI or E1/PRI termination used by the service provider [network provider] and how close it is to the MDF, the following tools and supplies are also needed. The T1 or PRI trunk is usually connected to the standard RJ-48C modular jack provided on the T1M or T1M-2 module. See page 161 for more information.

- For T1/PRI trunks terminated on RJ-48C jacks near the MDF or for E1/PRI trunks terminated on RJ-45 jacks near the MDF:
  - Four-pair, non-reversing, mod-to-mod line cords (one for each T1/PRI trunk installed).
  - For T1/PRI or E1/PRI trunks terminated on RJ-48C [RJ45 in Europe] jacks away from the MDF. See Figure 3 on **page 162**.
  - Eight-conductor modular jack [line jack unit] assemblies—two for each T1/PRI or E1/PRI span installed.
  - Four-pair, non-reversing, mod-to-mod line cords—two for each T1/PRI or E1/PRI span installed.
  - T1 or E1 cable—enough to extend from each telco RJ-48C [RJ45 in Europe] jack to the MDF. For cable specifications, see **page 87**.
- For T1/PRI or E1/PRI trunks terminated on an 66-type block:
  - Eight-conductor modular jack assemblies [line jack unit]—one for each T1/PRI trunk installed. See Figure 3 on **page 163**.
  - Four-pair, non-reversing, mod-to-mod line cords—one for each T1/PRI trunk installed.
  - T1 or E1 cable—enough to extend from each telco [network provider] termination to the MDF. For cable specifications, see **page 87**.

Planning Station Locations

Keep the following information in mind when planning station locations and phone types:

- **Phones:**
  - *IP*: Plan where IP phones are going to be located and whether a LAN port exists at that location. One LAN port is required for each phone. If a computer or other LAN device is plugged into this port (for example, a computer network cable), simply plug the IP phone into the LAN port, then plug the computer network cable into the LAN port on the IP phone.
  - *Digital:* The Digital Desktop Module (DDM-16), which can be installed only in the HX Controller chassis, and the Digital Expansion Interface (DEI) unit's Digital Endpoint Module (DEM-16), which can be used with either the HX Controller or CS Controller, can support various digital telephone models and the Single-Line Adapter (SLA). The SLA can be used to connect one or two single line phones, an OPX trunk, or—in the USA only—a DID trunk.
  - *Analog Single Line*: Analog single line phones include devices that operate on an industry-standard tip-ring/AC telephone circuit, such as DTMF phones, fax machines, answering machines, and playback devices. Playback devices are useful as hunt group overflow/announcement stations. For specifications, see **page 121**.

    Using the following interfaces, single line phones can be installed on the system:
    - Four-port Single Line Module (SLM-4)—for installation details, see page 199.
    - Eight-port Single Line Module (SLM-8)—for installation details, see page 230.
    - Single-Line Adapter (SLA)—for installation details, see page 211.
    - IP SLA—for installation details, see page 304.
    - Modem Data Port Module (MDPM) installed in a digital telephone—for installation details, see page 356.

- The single line ports on the back of the CS Controller chasis—for installation details, see page 142and page 159.

NOTE The RJ-14 jack(s) that support two single line (SL) ports are wired internally through a switch to the RJ-14 jack(s) that support two loop start (LS) trunk ports. In a system outage, the SL ports connect directly to the LS ports. This wiring design allows the customer to make one or more single line phones available that can be used to dial out on the LS trunks in the event of total system and/or site power loss. During normal system operation, the LS and SL ports can be programmed in DB Programming to function as desired.

- **PKMs (previously called DSS/BLF Units)**: Various Programmable Key Modules (previously referred to as Direct Station Select/Busy Lamp Field units) can be installed on digital telephones, depending on the model of phone. Each digital telephone supports up to four PKMs, and each MiVoice Office 250 supports a total of 20 PKMs. For installation details, see “Installing Other PKMs (DSS/BLF Units)” on page 346.
- **PKM 16 (previously called Mini-DSS Unit)**: Mitel 8520, 8560, and 8660 phones can connect directly to a PKM 16 (previously Mitel 8416). The number of PKM 16 units is not limited by system resources. For installation details, see “Installing a PKM 16 (Mini-DSS Unit)” on page 344.
- **Mitel 8664/8665/8668 wireless phones**: A NetLink SVP (SpectraLink® Voice Priority) server is required. For installation details, refer to the *SVP Server and Model 8664/8665/8668 Installation Manual*, part number 935.4521.

MIVOICE OFFICE 250 BASE SERVER CHASSIS

The following sections provide procedures for installing the newer HX Controller Base Server chassis and the original CS Controller chassis. The procedures appear in the order they should be performed for a new installation. Upgrade procedures and procedures for installing the Processing Server (PS-1), Digital Expansion Interface (DEI), expansion modules, and other optional equipment appear in separate sections.

Gathering Optional Equipment

Keep the following information in mind when gathering optional equipment, in preparation for installing a system:

- External voice mail systems (NuPoint Unified Messaging™) and applications software for integrated Voice Processing (such as voice mail, Automated Attendant, Hunt Group Announcement and Hunt Group Overflow, optional Fax-On-Demand, SMDR storage, etc.).
- CT Gateway for using System OAI applications on a network.
- Headsets for phones and handset amplifiers for single line phones.
- External music source.
- External paging speakers and amplifier(s). See the notice on **page 331**.
- Electronic equipment (alarms, lights, etc.) to connect to the night transfer relay.
- Fax machine. See **page 128**.
- Digital telephone PCDPMs for using Off-Hook Voice Announce (OHVA) features and for installing an MDPM. The serial port interface does *not* support serial data communication.
- Digital telephone MDPMs for connecting single line phones.
- 5310 conference units and Line Interface Modules.
- Optional wall-mount kits for the Base Server and DEI chassis.

Unpacking the Equipment

NOTICE

Electrostatic Discharge (ESD). The modules and components are static-sensitive. Handle the modules by the edges only and keep them in their protective bags until they are to be installed. Do not bend or touch the component pins or subject them to a static charge. When working with the modules, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the module during shipping. Any static charge (no matter how small) must be discharged from the body before touching the modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will not be replaced.*

Unpack the equipment and inspect the equipment chassis and all accompanying parts following these guidelines:

1. Check the parts against the packing slip and inspect them for damage. If any item is missing or damaged, contact Technical Support immediately. See “Technical Support Contact Information” on **page 1-370**.
2. Check all circuit module assemblies as follows:

a. Check each module to ensure there is no shipping foam or tape attached.

b. Inspect each module for shorted components.

c. Return all modules to their protective bags until they are ready to be installed in the chassis.

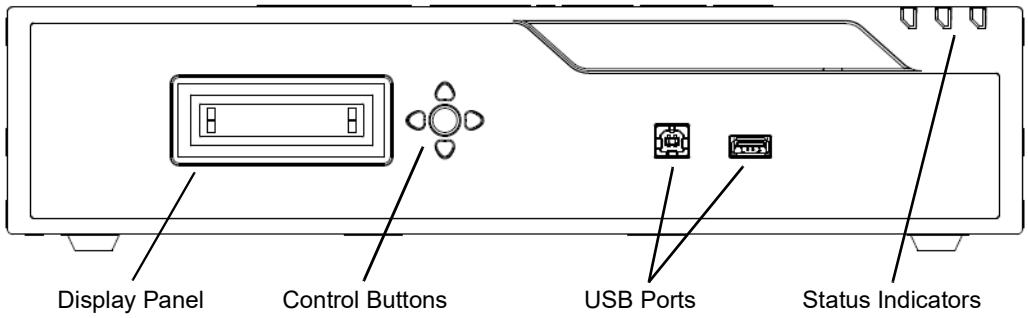
3. Check that the protective covers over the module bays are in place. These covers protect debris or other foreign matter from entering the system and possibly harming the internal components. These covers should remain in place until you are ready to install a module.

Installing the HX Controller Base Server Chassis

This section provides details on mounting the chassis, as well as connecting the power source, installing memory, setting up power fail transfer capability, and installing the optional door relay control.

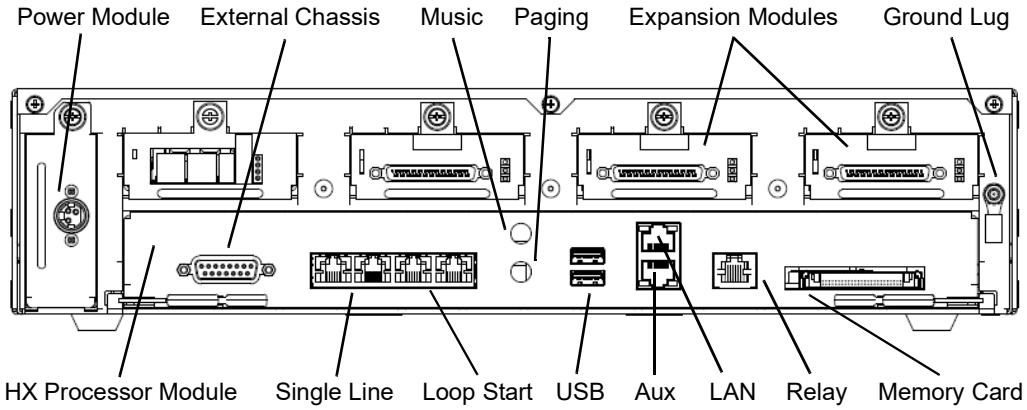
The following illustration shows the front of the HX Controller chassis.

Figure 10: HX Controller Front Panel



And this illustration shows the back of the HX Controller chassis equipped with an HX Processor Module, a T1 expansion module, and three digital expansion modules.

Figure 11: HX Controller Rear Panel



Mounting the Chassis

Locate the HX Controller chassis in a properly controlled environment and within 5 ft (1.5 m) of an isolated, dedicated, 105–125 VAC, 57–63 Hz, 15 A (North America) or 230-240 VAC, 50 Hz, 13 A (UK), single-phase commercial power source.

The chassis can be placed on a secure table, desk, or shelf, or it can be wall mounted or rack mounted as best suits the customer's needs.

To desk/table/shelf mount the chassis:

1. Locate the four desk-mount "feet" included in the chassis shipment.
2. On the bottom of the chassis, attach the four adhesive feet to the four corners as indicated by the silk-screened outlines.
3. Place the chassis on a stable, secure horizontal surface.

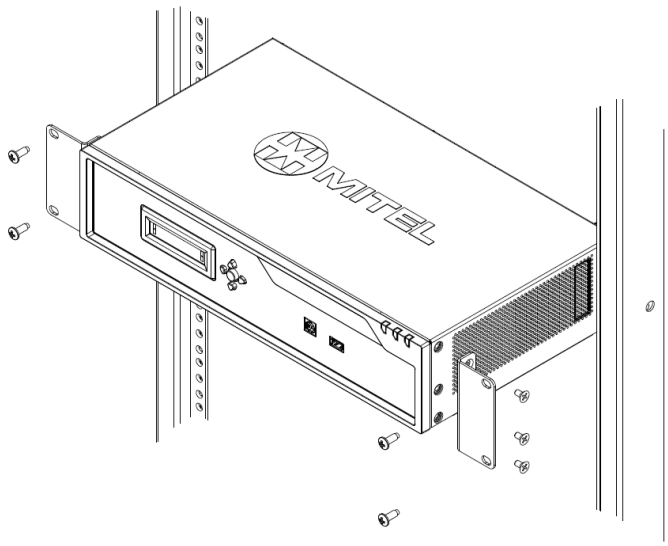
To wall mount the chassis:

Follow the instructions included in the optional standard wall-mount kit (part number 828.1771) or the optional expanded wall-mount kit (part number 50006504). For additional wall-mount options, contact the Mitel DataNet CommSource Division for more information.

To rack mount the chassis:

1. Locate the two L-shaped rack-mount brackets included in the chassis shipment.
2. Using **Figure 12** below as a guide, attach the two brackets to the left and right sides of the chassis using the screws included with each bracket.
3. Following the rack mount manufacturer's instructions and recommendations, attach the chassis securely to the rack.

Figure 12: Rack-Mount Bracket Installation



Connecting the Power Source

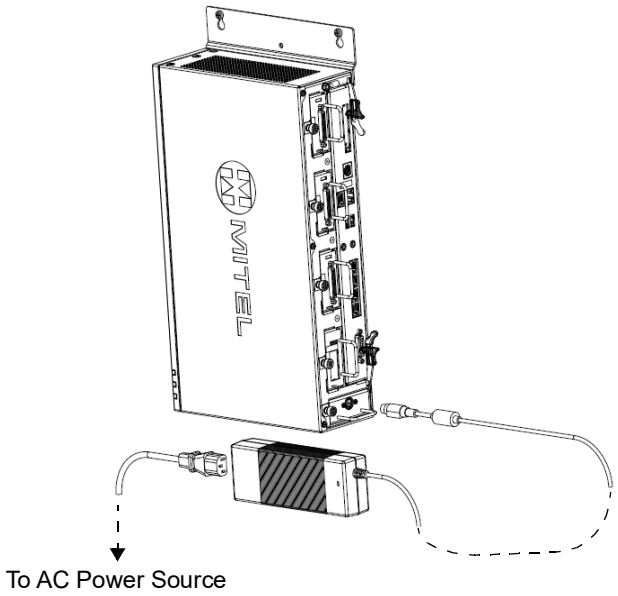
The HX Controller's external power supply unit must be connected to a properly grounded, isolated, and dedicated AC power source. Mitel recommends using appropriate surge/spike protection and an appropriate UPS unit.

Like with the chassis, the power supply unit can be placed on a secure table, desk, or shelf, or it can be wall mounted or rack mounted as best suits the customer's needs.

To attach the power supply unit:

1. Locate the power supply unit within five feet (1.5 meters) of the HX Controller chassis and within eight feet (2.4 meters) of an AC power source.
2. If rack mounting the unit, secure the unit to the rack using a suitable tie wrap or strap. If wall mounting the unit, refer to instructions included in the wall-mount kit.
3. Plug the male DIN connector on the end of the power supply unit's connector cable into the female DIN connector on the rear panel of the HX Controller chassis (see **Figure 13** below).
4. Plug the female C13 end of the AC power cord into the male C14 socket connector on the power supply unit.
5. Plug the other end of the AC power cord into an appropriate AC power source via a surge/spike protector and/or UPS unit. ***Do not*** plug in and turn on the UPS unit until you are ready to power on and boot up the system.
6. If using the UPS monitoring feature (see page 148 for details), plug the appropriate end of the USB cable included with the UPS unit into one of the USB-A ports on the rear panel of the HX Controller chassis. Then plug the other end of the USB cable into the appropriate connector on the UPS unit.

Figure 13: External Power Supply Connection



Setting Up Power Fail Transfer Capability

Like with the original CS Controller, the built-in loop start and single line ports on the rear of the HX Controller can be used to provide power failure bypass capability. If power to the system is interrupted, an internal electrical relay automatically connects two of the single line circuits (ports 1-2) with two of the loop start trunks (ports 1-2) to provide direct connection to local exchange dial tone. Under normal operating conditions, the circuits can be programmed and used the same as any other loop start or single line circuits. However, when system power fails, the circuits automatically connect, regardless of any programming that may have been done.

For installation instructions, see “Built-In Ports for WAN Fail Survivability” on page 158. The following tables provide the pinouts for the two six-pin RJ-14 single line jacks and two six-pin RJ-14 loop start jacks on the back of the HX Controller chassis.

Table 27: Built-in Single Line Jacks Pinout

Pin	Left Jack		Right Jack	
	Signal Name	Description	Signal Name	Description
1	No Connection		No Connection	
2	Tip 1	Single Line 4	Tip 3	Single Line 2
3	Ring 0	Single Line 3	Ring 2	Single Line 1
4	Tip 0	Single Line 3	Tip 2	Single Line 1
5	Ring 1	Single Line 4	Ring 3	Single Line 2
6	No Connection		No Connection	

Table 28: Built-in Loop Start Jacks Pinout

Pin	Left Jack		Right Jack	
	Signal Name	Description	Signal Name	Description
1	No Connection		No Connection	
2	Ring 1	Loop Start 2	Ring 3	Loop Start 4
3	Tip 0	Loop Start 1	Tip 2	Loop Start 3
4	Ring 0	Loop Start 1	Ring 2	Loop Start 3
5	Tip 1	Loop Start 2	Tip 3	Loop Start 4
6	No Connection		No Connection	

Installing the Door Relay Control

The HX Controller chassis has a six-pin modular relay jack that can be used to unlock a door by dialing a feature code. See the *MiVoice Office 250 Features and Programming Guide* for details on using this feature. The jack is connected to an optional external door answering unit (door box) following the instructions included with the unit.

The following maximum voltage requirements apply to the door relay:

- 0.3A, 125VAC
- 1A, 30VDC

It is recommended that the door relay be used below the maximum voltage rating to drive another relay to drive the door opener.

The following table provides the pinout for the six-pin modular relay jack on the back of the HX Controller chassis.

Table 29: Door Relay Jack Pinout

Pin	Signal Name
1	No Connection
2	No Connection
3	Lock Relay Common
4	Lock Relay Normally Open
5	No Connection
6	Lock Relay Normally Closed

In the de-energized state, the Lock Relay Common contact (pin 3) has continuity with the Lock Relay Normally Closed contact (pin 6), while the Normally Open contact (pin 4) is open. When the relay is energized, the Lock Relay Common contact and the Lock Relay Normally Open contact have continuity, leaving the Lock Relay Normally Closed contact open.

NOTE Door Relay port contacts are rated at 500mA @60 Vac or Vdc peak.

Installing the CS Controller Chassis

To install the Base Server:

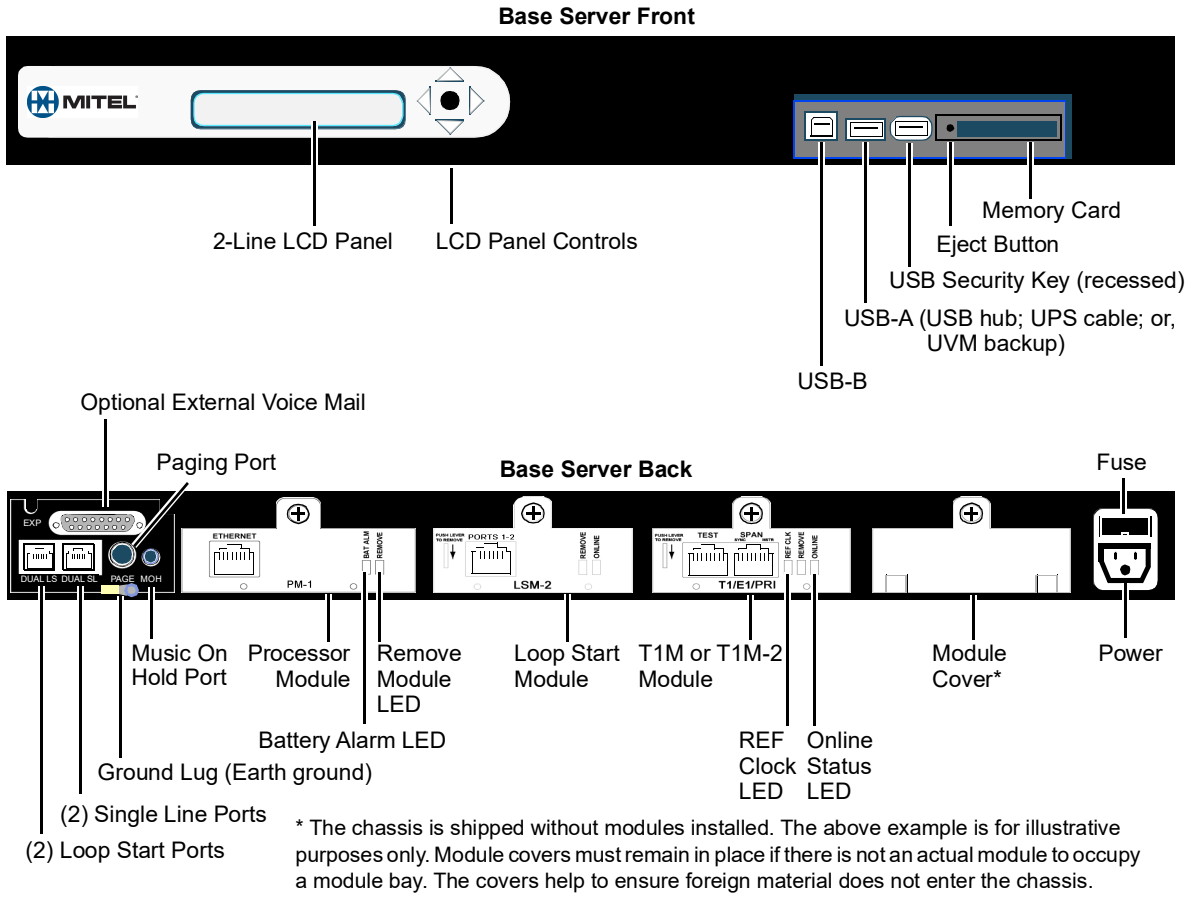
1. Install the chassis in the rack, then position the chassis and rack near the MDF backboard. Position the chassis at a height appropriate for reading the LCD panel. Also, allow room above the chassis for future expansion. For example, a voice processing system or Digital Expansion Interface (DEI) unit.

   The CS can be mounted in a standard rack, on a table, or on another secure, horizontal surface. Protective “feet” on the bottom of the chassis allow you to stack units, if not using a rack. There are external voice processing options available using NuPoint Unified Messaging (UM). These external voice processing systems can either be rack mounted (recommended) or placed on a horizontal surface in proximity to the CS.

2. *Do **not*** plug in the system AC power cable.

3. Continue to “Grounding the MiVoice Office 250 HX Controller” on page 146.

Figure 14: Base Server Front and Back Views



* The chassis is shipped without modules installed. The above example is for illustrative purposes only. Module covers must remain in place if there is not an actual module to occupy a module bay. The covers help to ensure foreign material does not enter the chassis.

Wall-Mounting the Base Server

Refer to the instructions included with the wall-mount kit (part no. 828.1692) to wall-mount the Base Server chassis.

Grounding the MiVoice Office 250 HX Controller

NOTICE

For proper grounding methods, refer to the *System Equipment Grounding Guidelines*, part number 835.2845. See page 133 for additional AC power requirements.

With a U.S. system, the MiVoice Office 250 HX Controller must be plugged into a UPS and the UPS plugged into a properly grounded AC power source. Some effective grounding connections include the steel frame of the building in which the system is installed, a grounding rod that has been driven a minimum of 8 ft (2.4 m) into the earth, or a copper—***not*** PVC—cold-water pipe.

With a European system, the equipment chassis must be plugged into a properly grounded AC power source. The Krone field should also be hard-wired to the building earth.

In addition, all optional IP Single-Line Adapters, trunk gas discharge tubes, and UPS battery compartments that are used must be properly grounded to the chassis AC outlet. The Mitel DataNet CommSource division or Mitel Europe sells a surge/spike protector that is equipped with a built-in ground lug just for this purpose.

The grounding wire should be 10AWG [diameter 6mm] or larger. To provide RFI/EMI protection only, it should not exceed 25 ft (7.5 m). To provide both RFI/EMI and lightning protection it should not exceed 10 ft (3 m).

For additional information, refer to the *System Equipment Grounding Guidelines*, part number 835.2845.

To ground the system:

1. Ensure the system AC power source is properly grounded. do ***not*** defeat (or leave floating) the ground wire in the AC power cord.
2. For a U.S. system only, install an AC surge/spike protector with a built-in ground lug and connect it to:

a. The ground lug (indicated by the symbol ⏚) on each Single-Line Adapter, if used.

b. The ground lug on the back of the chassis, as shown in Figure 14 on **page 144**.

c. Battery compartments, if used.

d. Trunk gas discharge tubes, if used.

3. Continue to “Base Server Modules” on **page 1-180**.

If AC power-related problems appear on the system while in operation, one of the following three methods can be used to test the ground.

- Use a digital multimeter to measure the difference of potential between the equipment cabinet ground point and the power neutral, and between the cabinet ground point and the third wire ground. If the voltage measurement exceeds 0.5 V, noise may develop on the system. If this occurs, call an electrician.
- Use a reliable earth-ground testing device to test the ground point. An earth-ground tester should be available through the local power company. Look for a reading of 5 ohms or less.

- For a U.S. system only, unplug the AC power cord from the outlet and insert a ground isolation plug into the outlet. Plug the AC power cord into the ground isolation plug. If the problem ceases, call an electrician. When finished, remove the ground isolation plug to restore ground protection.

Equipment to Support UPS Monitoring

This section describes the installation of Uninterruptible Power Supply (UPS) units that support various MiVoice Office 250 chassis. See “Compatible UPS Units” on **page 1-63**.

The UPS connects directly to a chassis USB port, and you should use only the USB cable provided by the UPS manufacturer. Product validation testing has determined that non-original equipment manufacturer (non-OEM) cables interfere with UPS Monitoring feature operation.

The UPS is connected to a USB-A port on the Base Server. Because only one USB port exists on the CS Controller Base Server, if the port is occupied with a flash drive, it must be removed. The use of a USB hub to increase the number of ports is also supported.

If installing an optional Processing Server (PS-1), the UPS must be connected to an available USB port on the PS-1. The UPS must connect to the server running call processing (CP) software in order for the UPS Monitoring feature to operate correctly.

To avoid electrically induced performance problems, all associated MiVoice Office 250 equipment should be sourced from the same AC phase and ground reference. To assure that the electrical source is the same, plug MiVoice Office 250 equipment into the same UPS unit. A common electrical source is most critical with the Base Server and associated Digital Equipment Interface (DEI) units.

Because the PS-1 is IP networked, a common electrical source is not as critical. However, if wattage and outlets are available, the PS-1 should be maintained on the same UPS. The PS-1 does not require its own UPS unit.

IMPORTANT	Comply with the following electrical requirements when installing or upgrading MiVoice Office 250 equipment: • Plug MiVoice Office 250 equipment into the same UPS unit, in particular the Base Server and associated Digital Equipment Interface (DEI) units. If wattage and outlets are available, the PS-1 should be maintained on the same UPS. • The UPS unit must be plugged into an isolated, dedicated circuit providing required voltage, current, and grounding specifications. • The power rating of the UPS unit must exceed the combined power ratings of all MiVoice Office 250 equipment the UPS supports.

Installing UPS Monitoring

Use the following procedure to install a UPS and activate UPS Monitoring on systems running v2.2 or later software.

To install a UPS and activate the UPS Monitoring feature:

1. Upgrade to the latest MiVoice Office 250 software release. See “Network Environment Requirements” on **page 1-307**.
2. The UPS Monitoring flag, Enable Shutdown on Low Battery, is set to **Yes** by default. Clearing this flag prevents a system from shutting down when a low battery event is received from the UPS unit.

3. Plug the system into the UPS. If a PS-1 is installed, both the Base Server and the PS-1 are assumed to be plugged into the same UPS. If the UPS unit provides outlets for both Surge Protection and Back Up Battery, use only the outlets for Back Up Battery.
4. Plug the UPS monitoring cable into the USB port of the system.
5. Do either of the following, according to your system type:
   - In a Baser Server system only that is not equipped with a compatible USB hub, remove the USB flash drive. The flash drive can be swapped with the UPS cable at a later time for backup purposes, but the UPS cable must be reconnected after backup for correct functionality of the UPS Monitoring feature.

     If the system is equipped with a compatible USB hub, voice mail can be backed up to a flash drive at the same time the UPS Monitoring feature operates.
   - In a system equipped with a PS-1, the UPS monitoring cable is plugged into an available USB port on the PS-1. The system then detects the device and an entry appears in Message Print. Even though the Enabling flag is set to **Yes** in DB Programming, UPS Monitoring functionality does not begin until a supported device has been detected.

NOTES

Consider the following system characteristics:

- Additional USB power events such as On Battery or Low Battery are also displayed in Message Print. If necessary, the option to shutdown the system on a low battery event can be disabled in DB Programming.
- Depending on the UPS vendor and model, the UPS itself may have additional configuration options such as low battery threshold. If available, the UPS configuration can be modified by loading the manufacturer's software on a PC, connecting the UPS to the PC, and configuring it directly. Any configuration of the UPS should be done prior to connecting the MiVoice Office 250 to it.

Installing the USB Security Key and Compact Flash-Type Memory Card

Locate the security key and compact flash-type memory card that came with the system. (With v3.2 or later software, the Base Server chassis can use a serialized compact flash-type card instead of a USB security key. See page **page 45** for details.) The memory card contains system software files and the applications necessary to operate the system. There are no application licenses on the security key; the key indicates that the system is able to download the corresponding license it needs to operate. For additional information about the USB security key and compact flash-type memory card, see the **Specifications** and **Product Description** chapters.

NOTICE

Version 3.x requires a minimum 512 MB compact flash-type memory card. Version 4.0 or later requires a minimum one GB compact flash-type memory card. The 512 MB card is not supported with v4.0 or later.

NOTICE

Possible Memory Card Corruption: By adhering to the following precautions you can help prevent corruption of data on the compact flash-type memory card:

- Do ***not*** remove or install the compact flash-type memory card while the system is up and running or power is otherwise supplied to the Mitel 5000 chassis, as this may damage the memory card. Shut down the system using the LCD panel and unplug the power cord before removing or inserting the memory card.
- Using any compact flash-type memory card other than one provided by Mitel is *not* supported.
- Do ***not*** place the compact flash-type memory card into any other PC. Doing so may corrupt the file system and software.

To install a USB security key and compact flash-type memory card:

1. Using the illustration on the next page as a guide, insert the USB security key in the USB port labeled “KEY.” Write down the key number (for example, on a sticker you can place above the key location on the chassis).

   The security key has changed over time. Newer keys are shorter than the previous security key. **Table 30** details the differences and compatibilities of the keys with the MiVoice Office 250 chassis.

Table 30: USB Security Key Port Compatibility

USB Security Key	Chassis	Compatibility
New key[a]	Original CS chassis[b]	Insert new key into a non-recessed port (USB-A). **Do not** insert into port labeled KEY.
	Newer CS chassis[c]	Insert into any USB port.
Existing key[d]	Original CS chassis	Insert into port labeled KEY.
	Newer CS chassis	Insert into any USB port.

a. Part number: 841.0892
b. Recessed USB port labeled “KEY”
c. No recessed USB ports.
d. Part number: 841.0219

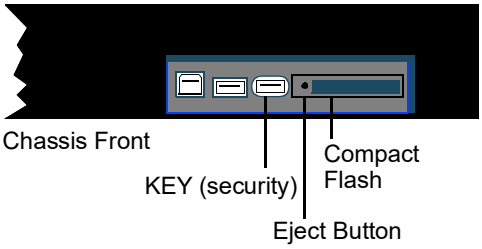
NOTICE

The system constantly verifies that the USB security key is in place. Do ***not*** remove the security key while the system is running. An alarm (122) is issued if the system detects a missing security key (the key is recessed into the chassis to decrease the possibility of its being dislodged and losing the connection). If the system detects a missing security key, it will invalidate the license in 30 minutes and reset the system in 4 hours. If the system is booted without the key in place, you will need to reload the license file in DB Programming, if it was previously loaded.

Also, If for any reason you need to replace the USB security key, make sure you upload the latest license file *before* replacing the key. Attempting to upload a license file *after* changing the USB security key results in an error and the system will not function.

2. Insert the compact flash-type memory card into the bay labeled COMPACT FLASH, as shown in the CS Controller example to the right.

3. Continue to “Verifying and Securing Cable Connections” on page 165.



Laying Out Cable Runs

Comply with the following the following guidelines when running cable:

- If necessary, run the appropriate network cabling to each IP phone location, or check that a LAN port exits at each location. For cabling requirements, see “Cabling to Phones” on **page 1-65**.
- If desired, two- or three-pair twisted cable and four- or six-conductor modular jacks [line jack units] may be used for *all* digital and analog phone connections. This allows the various types of phones to be easily interchanged as needed.

> **NOTE** If three-pair cable is used with a U.S. system, extra terminal blocks and the use of cross-connect wiring techniques are required.

- Install proper type cable for the application according to the National Electrical Code and local building codes [BS 6701 in Europe].
- Avoid cable runs parallel to fluorescent light fixtures or AC lines not in conduit. If these obstacles are unavoidable, run the cables across them at right angles.
- Do ***not*** run cables inside electrical conduit already occupied by AC power cable. To do so is a violation of the National Electrical Code, or BS 6701.
- Do ***not*** run cables near equipment with electric motors or through strong magnetic fields, such as those generated by large copy machines, arc welding equipment, heavy motors, etc.
- Do ***not*** place station cables where they can be stepped on or where they can be rolled over by office furniture.
- Do ***not*** exceed the loop limit measurements for the station cable lengths, using 24 gauge (AWG) wire.

For additional information about laying out cable runs for digital expansion installations, see “Optional System Equipment” on **page 1-328**.

Mounting and Cabling the Main Distribution Frame (MDF)

The MDF is the point at which the station instruments, trunks, and auxiliary system equipment are connected to terminal blocks [Krone blocks], or in the case of T1/PRI or E1/PRI trunks, modular jack [line jack unit] assemblies are used instead. These terminal blocks [Krone blocks] connect to the circuit modules in the CS Controller chassis. Connections *must* be made carefully and securely with solid copper wire. Do *not* use stranded, or speaker, wire to connect phones to the system.

With a European system each module is normally supplied with 2-meter cable, but 4-meter cables are available by special order. These cables are supplied with the appropriate number of Krone blocks for the module type.

To simplify the MDF installation instructions, this manual outlines basic one block-to-one module installation using bridging clips. However, it is quite acceptable, and in some situations more functional, to use additional blocks, then cross connect between them.

To assemble the MDF:

1. Mount a 4 ft x 6 ft (1.2 m x 1.8 m), ¾-inch (2 cm) plywood backboard at the MDF location. Depending on the number of stations and trunks to be installed, a larger backboard may be necessary.
2. With a U.S. system, attach the 66M1-50-type terminal blocks to the plywood backboard. With a European system, attach the Krone blocks to the plywood backboard. This includes blocks for connecting CO trunks.
3. If installing any T1/PRI and/or E1/PRI trunks, attach 8-pin, 8-conductor modular jack assemblies to the MDF as required. For information on installing T1/PRI or E1/PRI trunks, see **page 161**.

Connecting CO Trunk Cables to the MDF

This section provides information on connecting Central Office (CO) trunk cables for the LSM-2 and LSM-4 modules and the built-in loop start and single line ports. The built-in loop start and single line ports provide CO trunk capabilities to the single line phones in addition to providing the power-fail bypass functionality.

Connecting Trunks to LSM-2 Module Ports

For this procedure, see Figure 15 and Figure 16 on **page 155** as a guide.

NOTICE

Mitel will continue to support existing LSM-2s in the field, but new modules are no longer sold. For connecting loop start trunks to the MiVoice Office 250, use the LSM-4 module, part number 580.2304, or the T1/E1 module, part number 580.2700.

To connect LSM-2 Module CO trunks:

1. Cable the dual loop start port (CO trunks) using the pinouts shown in Figure 15 on **page 154**.
   - Use a patch panel at the rack and adapters between the patch panel and the Mitel 5000. Use the fewest modular jacks and adapters possible, and do *not* attach any adapters to the back of the chassis; clearance issues prevent them from attaching securely. If using a 6-pin,

6-conductor patch panel, make the appropriate adjustments when using the adapters. You can also build your own 6P/6C to 8P/8C patch cables, if necessary.

- If using 4- or 6-conductor modular line cord, limit the length to 6 ft maximum. In general, modular line cords are not twisted pair and could introduce crosstalk between line 1 and line 2 at longer lengths.

NOTICE

Equipment Damage Hazard. Use adapters when connecting any 6-pin plug to an 8-pin jack. Although a 6-pin plug will fit into an 8-pin jack, the plastic on the 6-pin plug housing will damage pins 1 and 8 on the 8-pin jack. Not using adapters will increase the likelihood of failure (to pins 1 and 8 in the 8-pin jack) should that jack need to be used with an 8-pin plug in the future. Various adapter types are available based on how you choose to connect to the patch panel.

2. Create the split to create two loop start port connections:
   - In Figure 15 on **page 154**, the split to achieve two loop start ports occurs at the patch panel. This is the recommended option.
   - In Figure 16 on **page 155**, the split to achieve two loop start ports occurs at the MDF using modular jacks.

Figure 15: Cabling Option 1 for the LSM-2

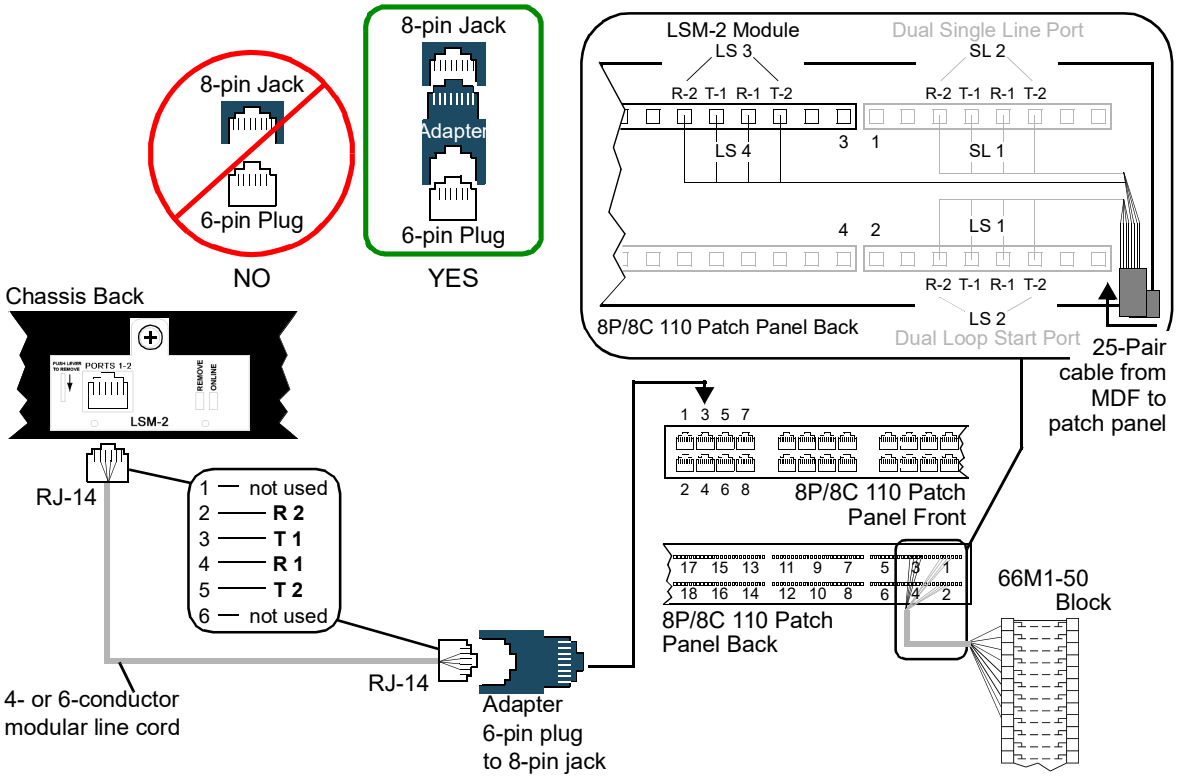
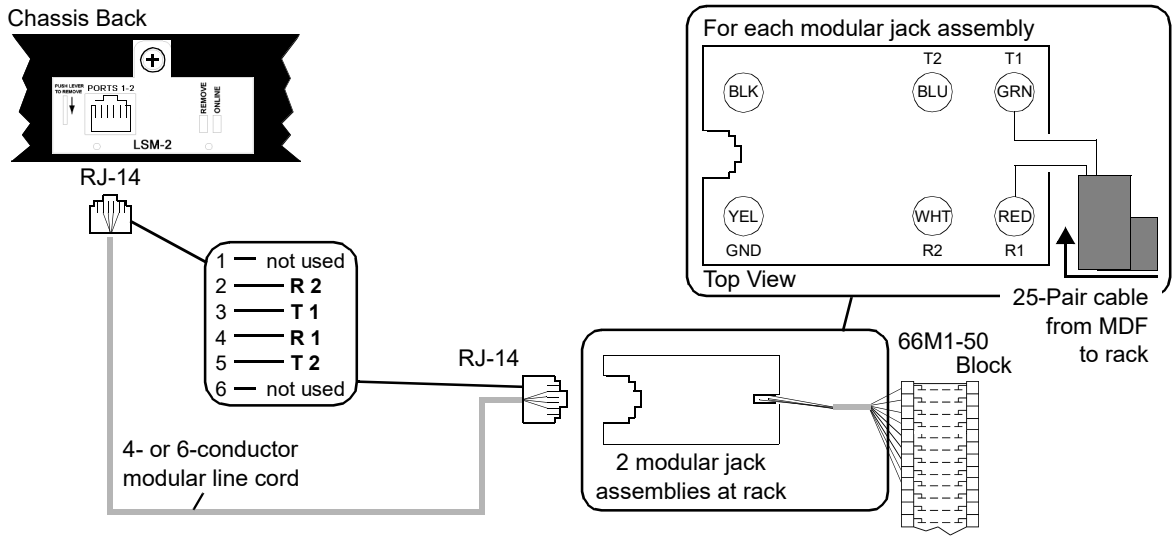


Figure 16: Cabling Option 2 for the LSM-2



Connecting Trunks to LSM-4 Module Ports

This section provides information on connecting CO trunk cables for the LSM-4 module. For this procedure, see Figure 15 and Figure 16 on **page 155** as a guide.

To connect LSM-4 Module CO trunks:

1. Cable the four-port loop start module port (CO trunks) using the pinouts shown in Figure 15 and Figure 16 on **page 155**.
   - Use a patch panel at the rack and adapters between the patch panel and the Mitel 5000. Use the fewest modular jacks and adapters as possible, and do not attach any adapters to the back of the chassis; clearance issues prevent them from attaching securely. If using a 6-pin, 6-conductor patch panel, make the appropriate adjustments when using the adapters. You can also build your own 6P/6C to 8P/8C patch cables, if necessary.
   - If using 4- or 6-conductor modular line cord, limit the length to 6 ft maximum. In general, modular line cords are not twisted pair and could introduce crosstalk between line 1 and line 2 at longer lengths.

NOTICE

Equipment Damage: Use adapters when connecting any 6-pin plug to an 8-pin jack. Although a 6-pin plug will fit into an 8-pin jack, the plastic on the 6-pin plug housing will damage pins 1 and 8 on the 8-pin jack. Not using adapters will increase the likelihood of failure (to pins 1 and 8 in the 8-pin jack) should that jack need to be used with an 8-pin plug in the future. Various adapter types are available based on how you choose to connect to the patch panel.

2. Create the split to create two loop start port connections:
   - In Figure 15 on **page 154**, the split to achieve two loop start ports occurs at the patch panel. This is the recommended option.

- In Figure 16 on **page 155**, the split to achieve two loop start ports occurs at the MDF using modular jacks.

Figure 17: Cabling Option 1 – LSM-4

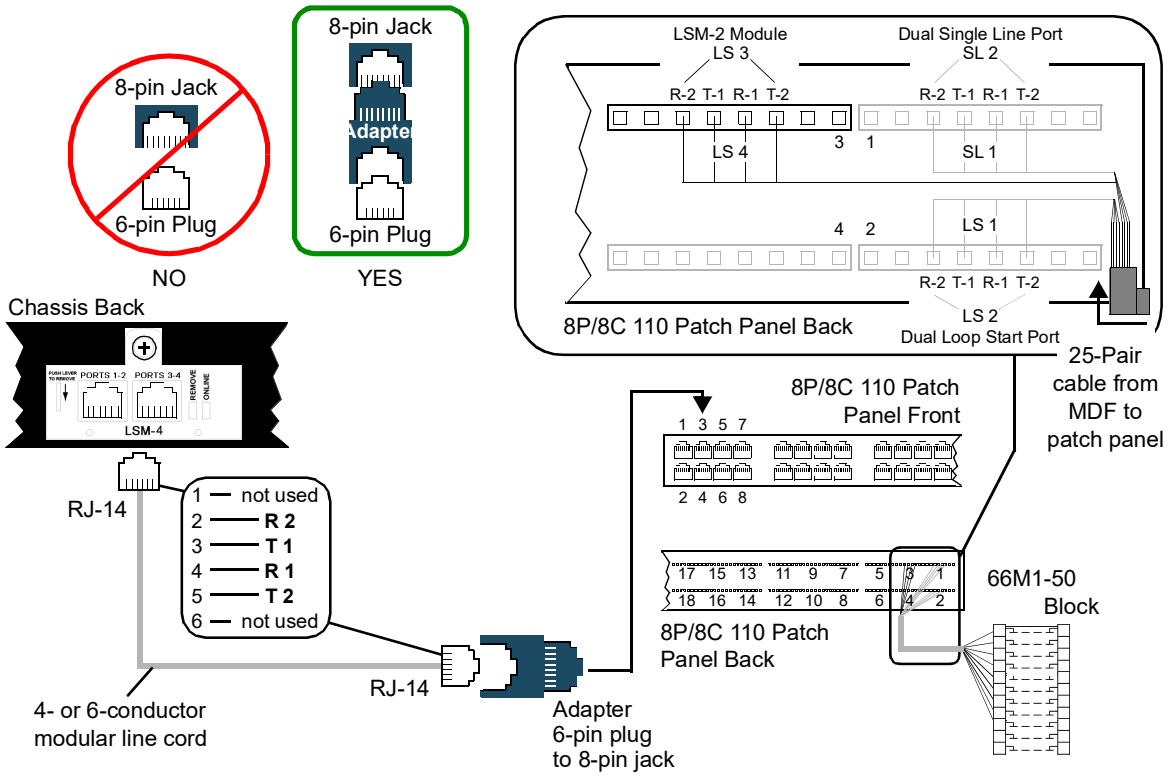
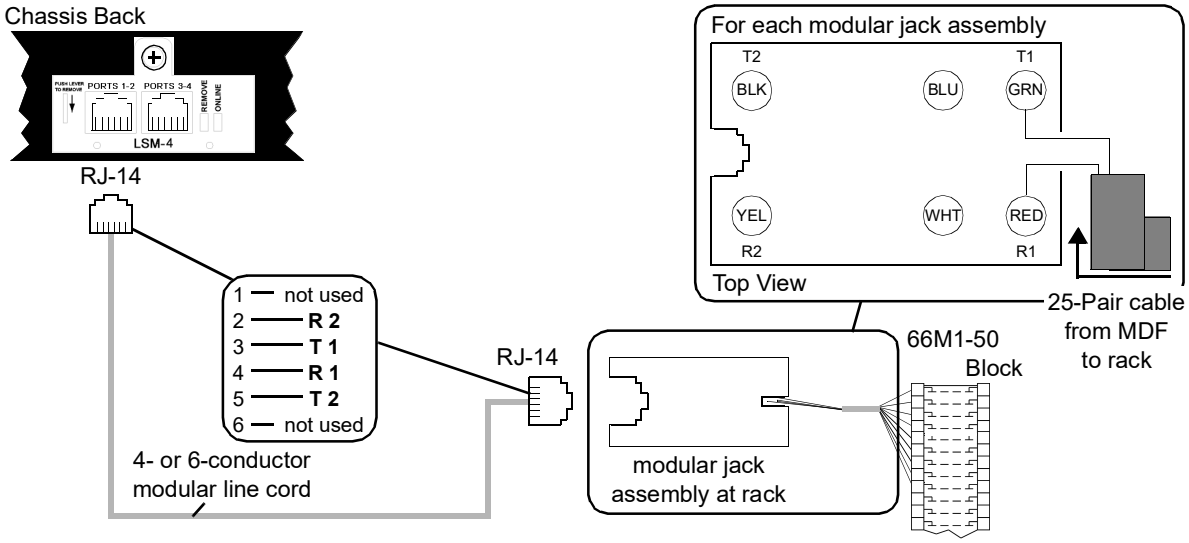


Figure 18: Cabling Option 2 – LSM-4



For additional lightning protection, install gas discharge tubes with silicon avalanche suppressors to ground (grounding rod or copper cold water pipe) on each CO trunk. This must be done external

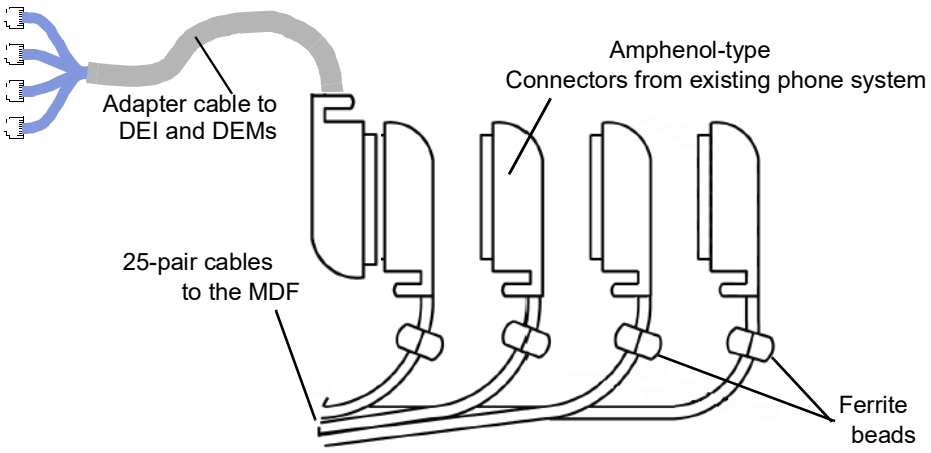
to the system. Each gas discharge tube is installed directly between the service provider company RJ-type block and the system CO/DID/OPX input. This protection should provide energy absorption and filter low-level surge potentials. Also, in areas with frequent occurrences of lightning, make the cable length between the service provider company RJ-type blocks and the gas discharge tubes at least 75 ft (22.5 m). The cable may be coiled as needed.

Connecting CO Trunks to a U.S. System

Connect the CO trunks and other module cables. This procedure describes additional information for installing digital telephones. Note that step 2 mentions two installation methods. Although the first method is more flexible, the second method can be used to minimize the number of terminal blocks required.

1. Cross-connect the CO trunks from the telephone company RJ-type blocks to the right side of the Mitel system blocks at the MDF. To provide additional lightning protection, see “Installing Lightning Protection” on **page 1-163**.
2. For a new installation, use enough 25-pair cable to run from the MDF to the modules. To help speed installation, a 15 ft (4.5720 m) cable with Amphenol connector is available, part number 813.1814.
   - *Replacing an Inter-Tel Axxess system:* Use the 15 ft Amphenol-terminated cable as appropriate. One end of the cable is terminated with a male Amphenol-type connector that will plug into a female Amphenol-type connector used for terminating Axxess system stations at the MDF. The other end of the cable is terminated with 4 prewired RJ-45 plugs for quick connection to the DEM-16s in the DEI chassis. *If necessary,* using standard electrical tape, attach two ferrite split beads, part number 808.1195, around each 25-pair cable just below the Amphenol-type connector, as shown in **Figure 19**. Ferrite beads may be necessary to help reduce EMI and RFI.

Figure 19: Example of Ferrite Bead Installation and Connection Types



3. Terminate the other end of each 25-pair cable on the left side of the corresponding cable termination block, and install bridging clips to complete the connections.

Connecting CO Trunks to a European System

Some trunks are connected using the fallback system. See “Maintenance Tasks” on **page 1-247**. The trunk connections are made to the designated Krone blocks.

Lightning Strike Protection Methods

For protecting the system against lightning strikes, attach the two ferrite split beads around each cable just below the Amphenol-type connector, as shown in Figure 19 on **page 157**.

For additional lightning protection, install gas discharge tubes with silicon avalanche suppressors to ground (grounding rod or copper cold water pipe) on each CO, DID, and OPX trunk. This must be done external to the system. Each gas discharge tube is installed directly between the telephone company (network provider) RJ-type block and the system CO/DID/OPX input. This protection should provide energy absorption and filter low-level surge potentials. Also, in areas with frequent occurrences of lightning, it is recommended that the cable length between the telephone company (network provider) RJ-type blocks and the gas discharge tubes be at least 75 ft (22.5 m). The cable may be coiled up as needed.

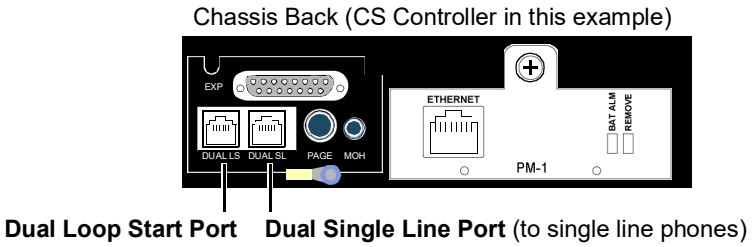
Built-In Ports for WAN Fail Survivability

The loop start ports and single line ports are built-in to the CS Controller to provide Wide Area Network (WAN) Fail Survivability through Power Fail Bypass circuitry. When not in WAN Fail Survivability mode, the ports support CO loop start trunks and single line phones in the system configuration.

The WAN Fail Survivability feature allows calls to be made from one more single line phones through the CO [branch exchange] during a network outage. A customer can dial out and receive calls at any time through these connections. But, the single line phones and loop start trunks connected to the built-in ports continue to operate during a network outage. For more information about WAN Fail Survivability, see the **Product Description** chapter.

RJ-14 jacks on the back of the CS Controller provide interfaces for the Loop Start Ports and Single Line Ports that make up the WAN Fail Survivability feature. See **Figure 20**.

Figure 20: Built-In Loop Start and Single Line Ports



For this procedure, see Figures 21 and 22 on **page 1-160** as a guide. For HX Controller platforms, see also “Setting Up Power Fail Transfer Capability” on page 142.

To connect the built-in loop start CO trunks and single line port:

1. Run the necessary amount of cable and install the phone you want to use (single line set) should power fail to the system.
2. Cable the single line and loop start ports.
   - Two-pair cables are required for each dual single line port and each dual loop start port.
   - Use a patch panel at the rack and adapters between the patch panel and the CS Controller chassis. Use the fewest modular jacks and adapters as possible, and do not attach any adapters to the back of the chassis; clearance issues prevent them from attaching securely. If using a 6-pin, 6-conductor patch panel, make the appropriate adjustments when using the adapters. You can also build your own 6P/6C to 8P/8C patch cables, if necessary.
   - If using 4- or 6-conductor modular line cord, limit the length to 6 ft maximum. In general, modular line cords are not twisted pair and could introduce crosstalk between line 1 and line 2 at longer lengths.

NOTICE

Equipment Damage: Use adapters when connecting any 6-pin plug to an 8-pin jack. Although a 6-pin plug will fit into an 8-pin jack, the plastic on the 6-pin plug housing will damage pins 1 and 8 on the 8-pin jack. Not using adapters will increase the likelihood of failure (to pins 1 and 8 in the 8-pin jack) should that jack need to be used with an 8-pin plug in the future. Various adapter types are available based on how you choose to connect to the patch panel.

In Figure 21 on **page 160**, the split to achieve two loop start and two single line ports occurs at the patch panel. This is the recommended option. In Figure 22 on **page 160**, the split to achieve two loop start and two single line ports occurs at the MDF using modular jacks.

Figure 21: Cabling Option 1 for the Dual LS and Dual SL

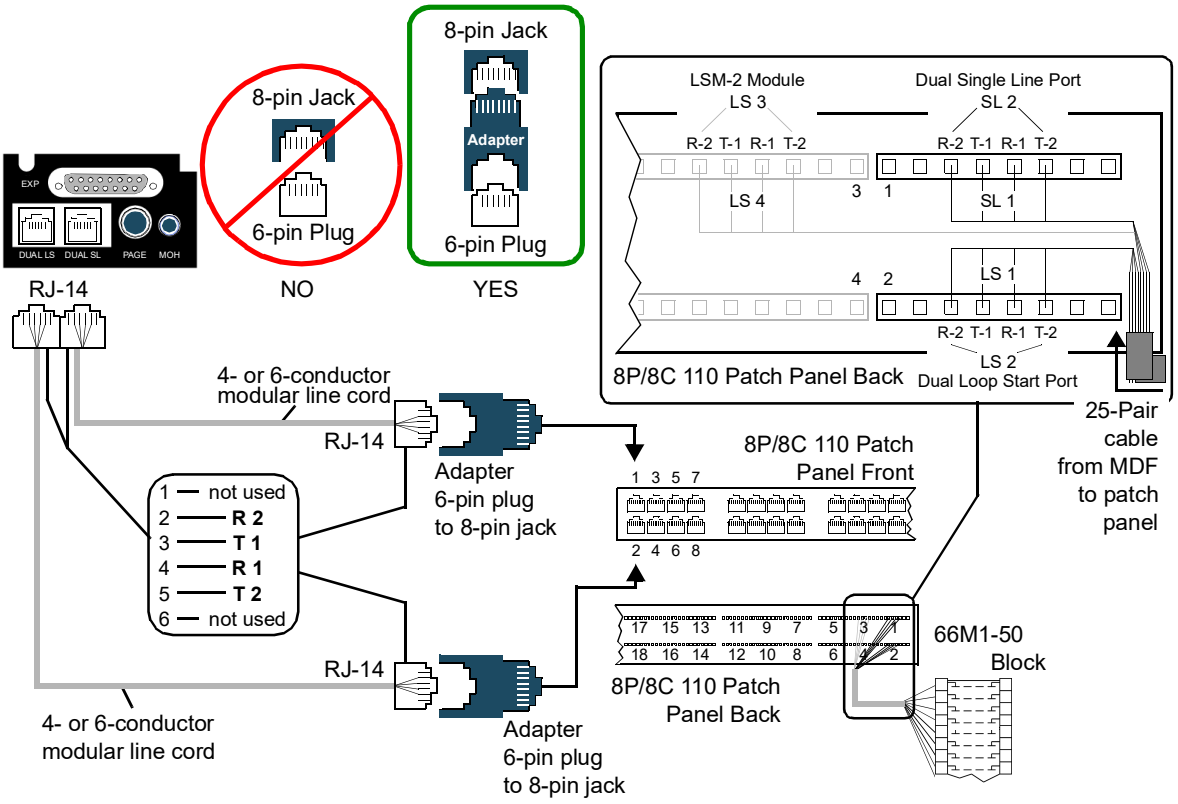
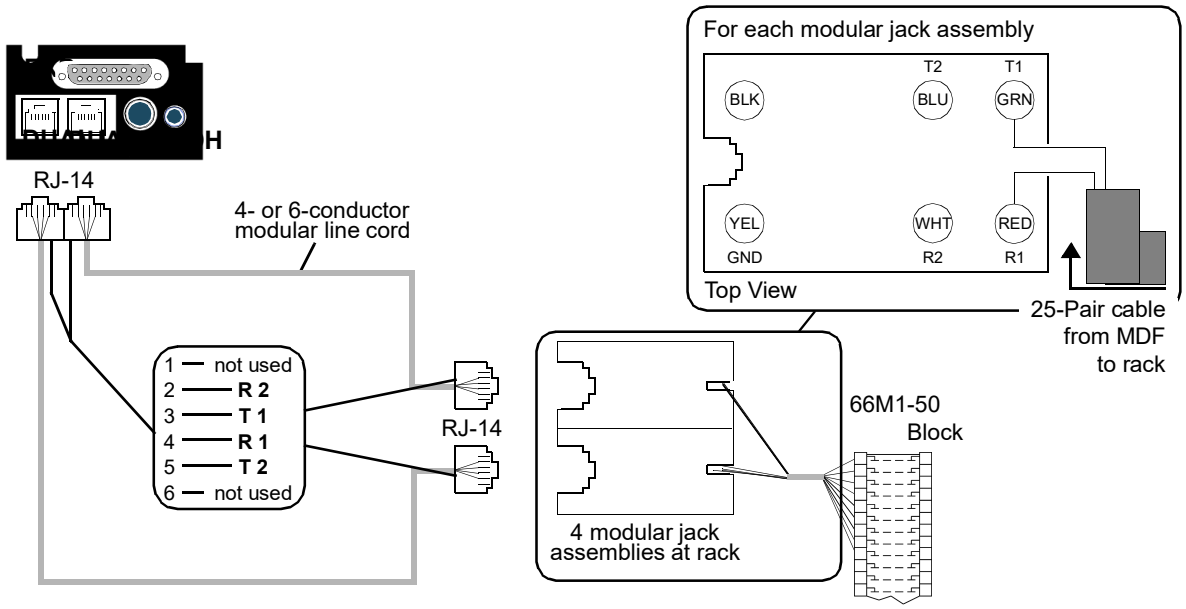


Figure 22: Cabling Option 2 for the Dual LS and Dual SL



Connecting T1/PRI and E1/PRI Trunks to the MDF

The installation procedure used to connect T1/PRI and/or E1/PRI spans at the MDF depends on the type of termination used by the service provider [network provider] and how close it is to the MDF. T1/PRI or E1/PRI trunks may be terminated on RJ-48C modular jacks [RJ45 line jack units] or on 66-type terminal blocks.

> **NOTE** Primary Rate Interface for ISDN service requires separate licensing.

Follow one of the termination methods (A, B, or C) outlined below and illustrated beginning on **page 162**. The T1/PRI trunk is usually connected to the standard RJ-48C jack provided on the T1M or T1M-2 module.

Method A – T1/PRI Trunks Terminated on RJ-48C Jacks Near the MDF

To install T1/PRI trunks terminated on RJ-48C jacks near the MDF:

1. Plug one end of a four-pair, non-reversing [straight through], mod-to-mod line cord into each service provider RJ-48C jack. The other end of each line cord will later be plugged into the jack on the corresponding T1M or T1M-2 module.
2. Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, transmission problems can occur.

Method B – T1/PRI Trunks Terminated on RJ-48C Jacks Away from the MDF

To install T1/PRI trunks terminated on RJ-48C jacks away from the MDF:

1. Mount one eight-conductor modular jack [line jack unit] assembly next to each service provider RJ-48C jack. For a diagram of the complete layout, see Figure 3 on **page 162**.
2. For each modular jack [line jack unit] assembly mounted in step **1**, mount a corresponding four-conductor modular jack [line jack unit] assembly on the MDF backboard.
3. Connect the modular jack [line jack unit] assemblies mounted in steps **1** and **2**.
   a. Run T1 cable between the corresponding modular jack [line jack unit] assemblies.

> **NOTE** If the service provider termination is more than 2000 ft (600 m) from the MDF, special T1 cable *must* be used (see **page 87** for cable specifications). For shorter distances, two lengths of standard two-pair voice-frequency cable (one cable for transmit and one cable for receive) may be used instead.

   b. Wire each end of the cables onto their respective modular jack [line jack unit] assemblies. See Figure 4 on **page 162** for a diagram of the wiring.
4. At the service provider termination, plug one end of a four-pair, non-reversing [straight through], mod-to-mod line cord into each RJ-48C jack and plug the other end into the corresponding modular jack [line jack unit] assembly next to it.
5. At the MDF backboard, plug one end of a four-pair, non-reversing [straight through], mod-to-mod line cord into each modular jack [line jack unit] assembly. The other end of each line cord will *later* be plugged into the jack on the corresponding T1M or T1M-2 module. Make sure the

connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, transmission problems can occur.

Figure 3: T1/PRI or E1/PRI Terminations from RJ-48C [RJ45 in Europe] Jacks

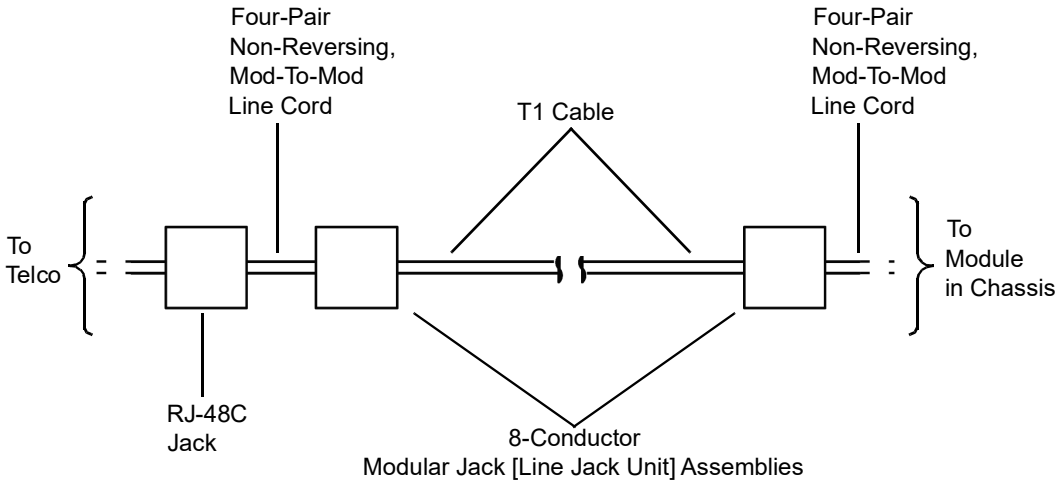
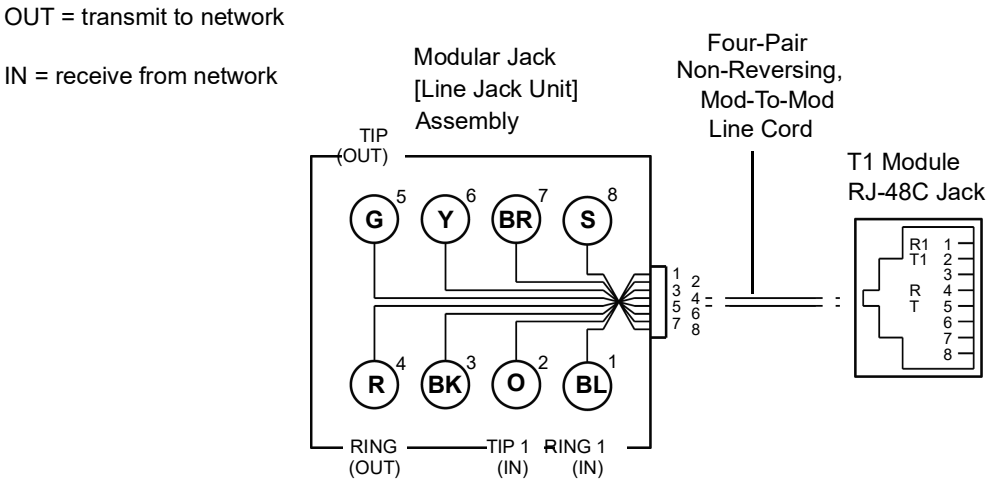


Figure 4: Modular Jack [Line Jack Unit] Assembly Wiring for T1/PRI Trunks



Method C – T1/PRI Trunks Terminated on a 66-Type Block

To install the T1/PRI trunks terminated on an 66-type block:

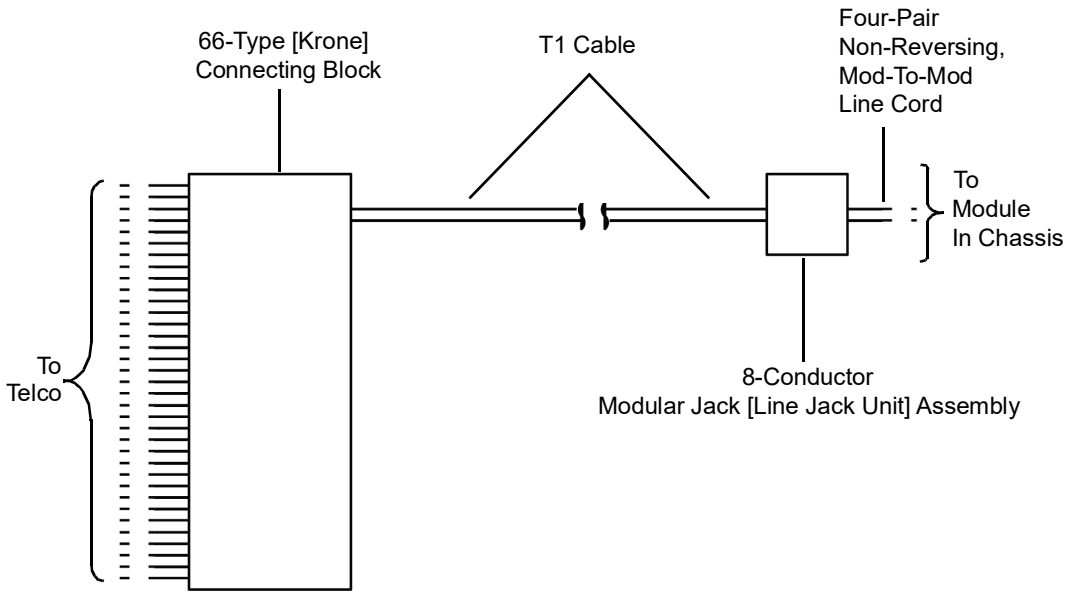
1. On the MDF backboard, mount one eight-conductor modular jack [line jack unit] assembly for each T1/PRI trunk termination on the service provider 66-type block(s). See Figure 3 on page 163 for a diagram of the complete layout.
2. Connect T1 cable between the service provider [network provider] terminations and the corresponding modular jack [line jack unit] assemblies as follows:
   a. Terminate one end of a four-pair cable on each modular jack [line jack unit] assembly. See Figure 4 on page 162.

b. Terminate the other end of the four-pair cable on the *right* side of the T1/PRI 66-type terminal block.

If the service provider [network provider] termination is more than 2000 ft (600 m) from the MDF, special T1 cable must be used. For cable specifications, see **page 87**. For shorter distances, two lengths of standard two-pair voice-frequency cable—one cable for transmit and one cable for receive—may be used instead.

3. Plug one end of a four-pair, non-reversing [straight through], mod-to-mod line cord into each modular jack [line jack unit] assembly, which was mounted in **step 1**. The other end of each line cord will *later* be plugged into the T1 jack on the corresponding T1M or T1M-2 module. Make sure the connectors on the ends of each line cord are properly attached. If any of the wires are cracked or broken, T1/PRI transmission problems can occur.

Figure 3: T1/PRI or E1/PRI Trunk Terminations from 66-Type Blocks



Installing Lightning Protection

For additional lightning protection, install gas discharge tubes with silicon avalanche suppressors to ground (grounding rod or copper, cold water pipe) on each CO, DID, and OPX trunk. This must be done external to the system. Each gas discharge tube is installed directly between the service provider [network provider] 66-type block and the system CO/DID/OPX input. This protection should provide energy absorption and filter low-level surge potentials.

In areas with frequent occurrences of lightning, check that the cable between the service provider [network provider] 66-type [Krone] blocks and the gas discharge tubes is at least 75 ft (22.5 m) long. The cable may be coiled as desired.

Connecting T1M or T1M-2 Modules between Network Nodes

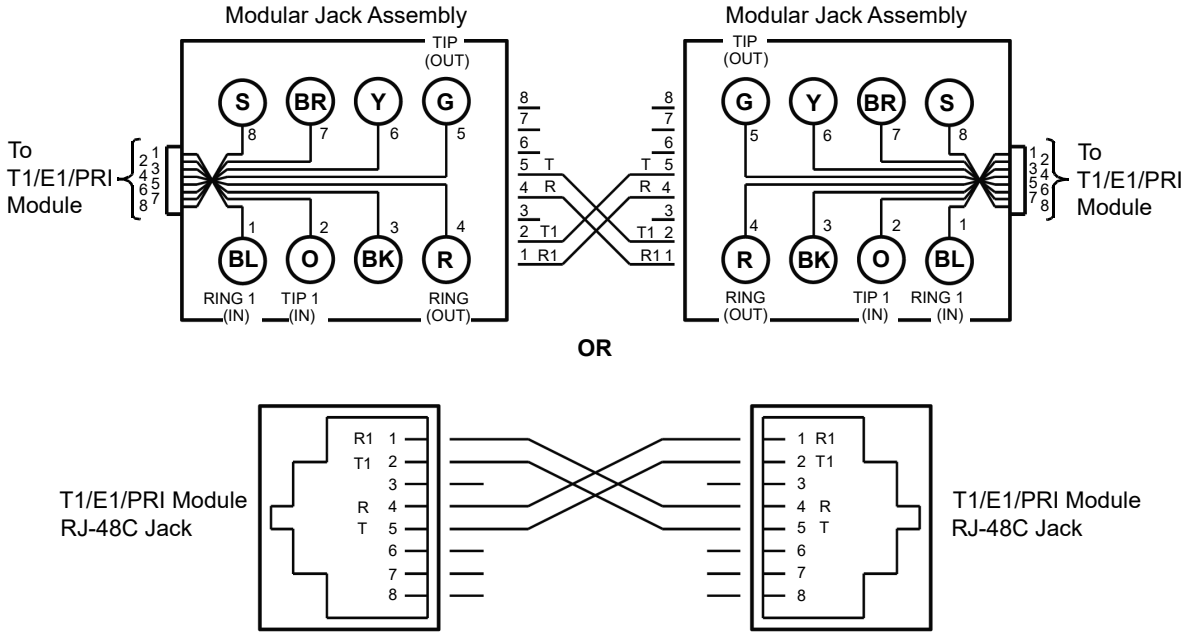
T1M or T1M-2 modules are used for connecting system nodes in a network. Each network span requires two T1M or T1M-2 modules, one for each end of the span. Point to point, clear channel T1

spans are used to connect the nodes. For each network span, order a clear channel, point to point T1 or E1 span from the service provider. Do not order PRI service, even though you will be connecting to T1M or T1M-2 modules.

T1M or T1M-2 modules can be used to connect two non-networking systems together using customer-provided cable. The systems may be located in separate buildings and the cabling between them may be as long as 6000 ft (1800 m), without repeaters.

When connecting two systems together, the line cord (or cabling) between the two T1M or T1M-2 modules must be wired so that the receive pair (tip 1 in and ring 1 in) on each module is connected to the transmit pair (tip out and ring out) on the other module, as shown in **Figure 4**. For information about programming various types of T1M or T1M-2 installations, refer to the *MiVoice Office 250 Features and Programming Guide*.

Figure 4: Connecting Two T1M or T1M-2 Modules Together



Verifying and Securing Cable Connections

At this point in the installation all cabling to the MDF, patch panel, rack, and phones should be completed, including all cables that terminate at the chassis modules.

To complete cabling:

1. Plug all applicable cables into their respective modules and chassis ports.
2. Install the LAN cable from the CS Controller or HX Controller RJ-45 LAN jack to a LAN connection. Plug the cable into a network switch, *not* a hub. (The LAN port is configured to auto-negotiate both speed and duplex mode. As such, the network switch port connected to this jack must also be configured to auto-negotiate speed and duplex mode.)

Setting Initial System Configuration

Upon initial power up, the system uses DHCP to establish a network connection. If the site is not running a DHCP server, you must provide site-specific IP addresses and other network information before you can begin programming the system in DB Programming. You can also program the database offline using the Offline Mode option in the System Administration & Diagnostics application.

NOTICE

To avoid system malfunctions observe the following rules when installing a MiVoice Office 250:

- DHCP is enabled by default to allow the system to become operational upon boot up. After the system has booted up, replace the DHCP address(es) with static IP addresses. If using a DHCP server, the IP address may change, dropping any CT Gateway and/or Voice Processing connections.
- IP Addresses, subnet mask, and gateways: it is not necessary that the CT Gateway be on the same LAN as the IP address and subnet mask of the system.
- Systems equipped with a Processor Expansion Card (PEC-1) have two IP addresses—one for the Processor Module (PM-1) and one for the PEC-1. Both IP addresses MUST be on the same subnet mask.

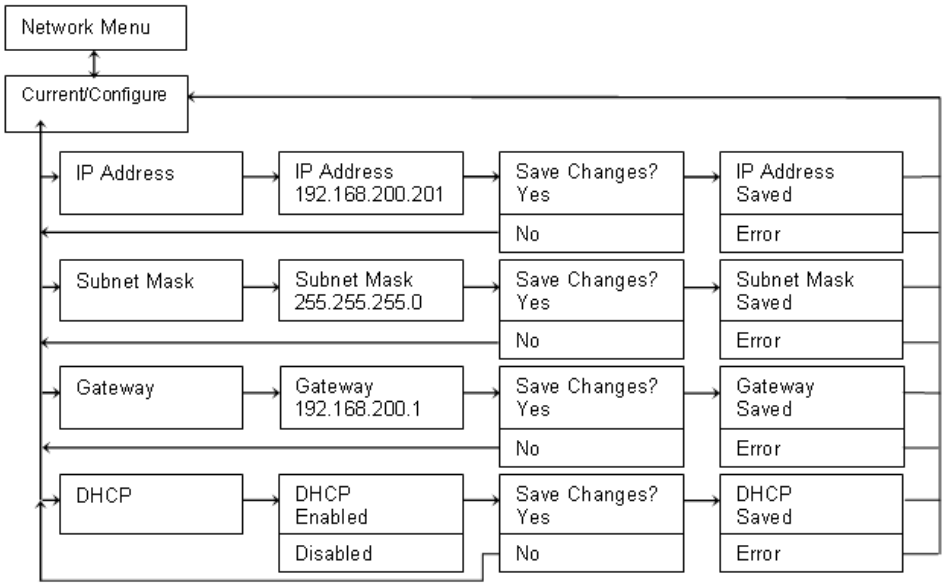
To complete the initial system configuration, use the Liquid Crystal Display (LCD) panel and controls on the front of the server. The LCD allows you to navigate various menus and options provided by the LCD application. For detailed information about the LCD application menus and options, see “Using the LCD Panel“ on **page 169**.

To set initial configuration options:

1. Obtain the required network information from the site administrator.
2. Power on the server. (See “Powering Up the System” on **page 1-247**.) The server IP addresses are assigned through DHCP.
3. From the idle menu (**MiVoice Office 250**), press the down arrow button to navigate to the Network menu.

   To navigate to and select menu options:

   a. Press the up or down arrow buttons to see the available options.
   b. When the appropriate option is displayed, press the center button to select it.
   c. Press the center button again to confirm the selection.
4. Press the down arrow button to configure the following:
   - **IP Address**: Use the static IP settings provided by your network administrator.
   - **Subnet Mask**: Use the static IP settings provided by your network administrator.
   - **Gateway**: Use the static IP settings provided by your network administrator.
   - **DHCP**: Disable DHCP.



To edit an IP address:

a. Press the left arrow when the cursor is blinking on the left digit (of the address you want to edit). This deletes the address.

b. Use the up and down arrow buttons to change the digits.

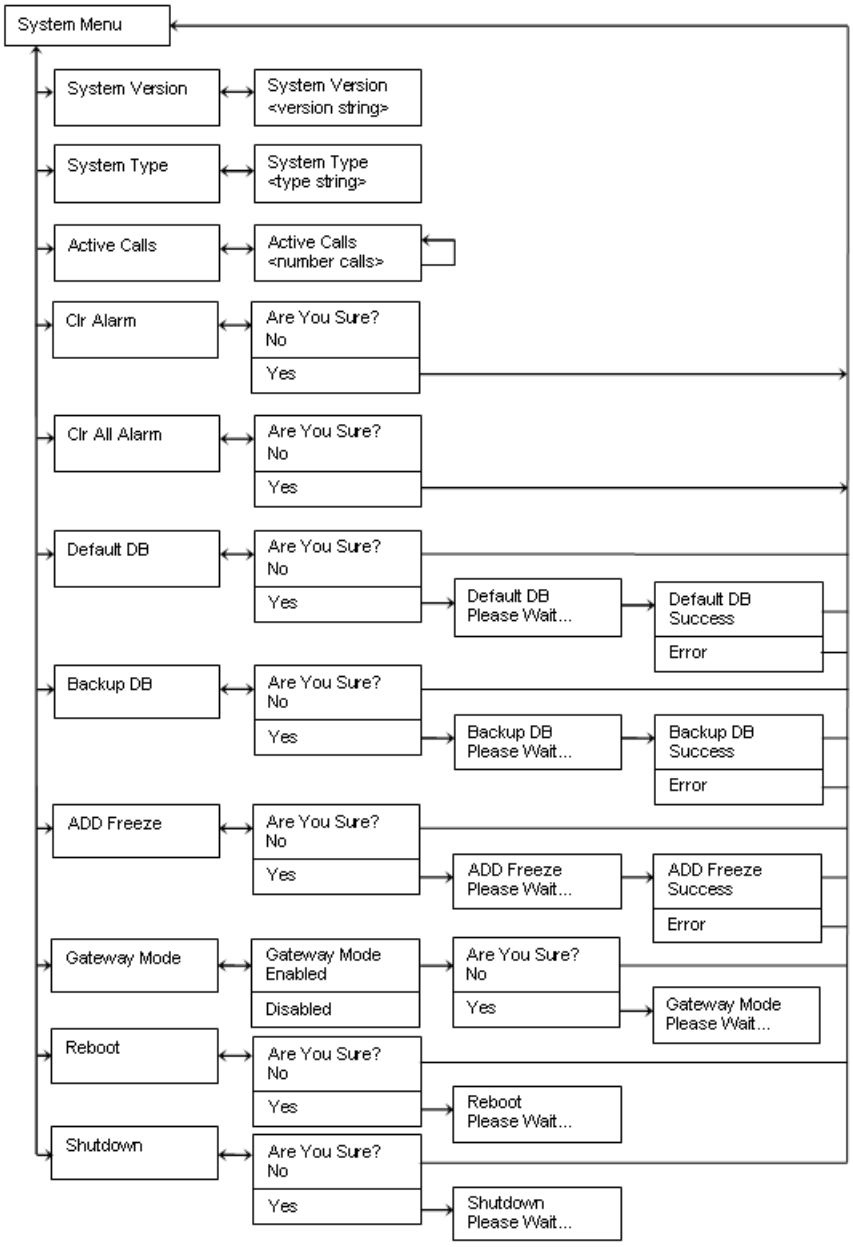
c. Press the center button to save your selection. The **Save Changes?** prompt appears.

d. Press the center button again to confirm the selection.

You have 15 seconds to make your selection or the LCD returns to the default idle display, **MiVoice Office 250**.

5. After you change the system's network settings, return to the idle menu and then reboot the system as follows:

a. From the idle menu (**MiVoice Office 250**), press the down arrow button to navigate to the System Menu.



b. Navigate to the **Reboot** option and then press the center button to select it. The **Are You Sure?** prompt appears.

c. Press the down arrow button to select **Yes**. Wait while the system reboots.

6. After the system reboots, verify that the front of the LCD reads **MiVoice Office 250,** and then do one of the following:

- Refer to the *MiVoice Office 250 Features and Programming Guide* for information about configuring other aspects of the system and IP phones. If you made changes to DB Programming while making changes to the LCD panel, restart DB Programming.
- Continue to “Installing Mitel 86xx IP Phones” on **page 1-265**.

Using the LCD Panel

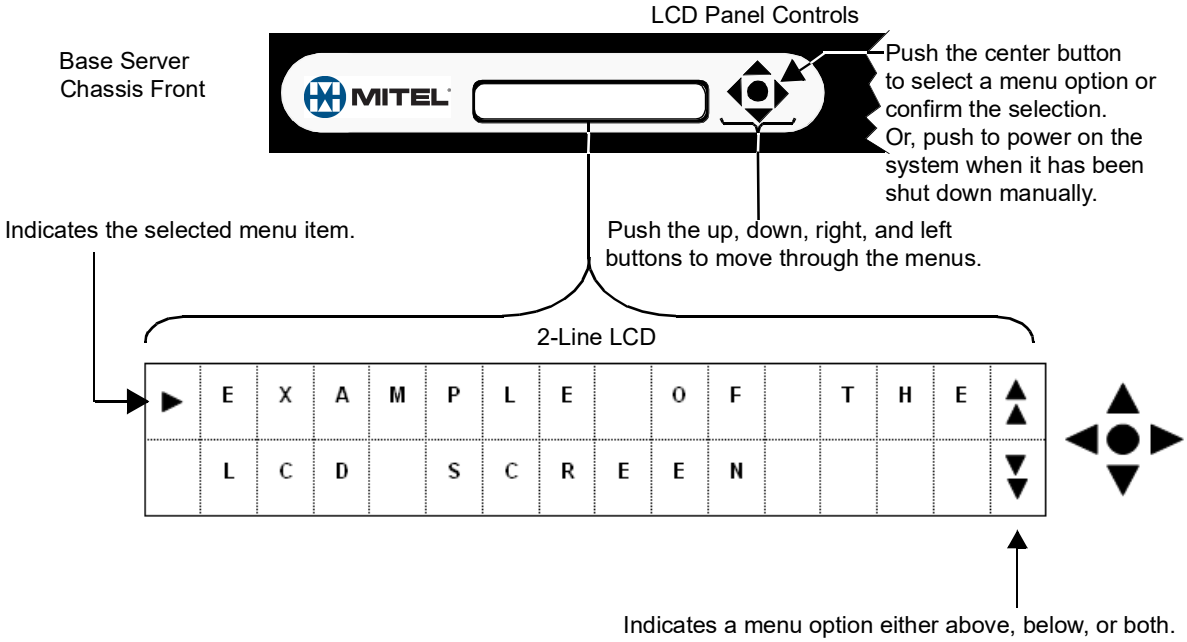
The Base Server chassis is equipped with an LCD panel that displays system messages, menus, and options. This section describes the LCD panel hardware, the application software, the system messages, and the menus and options provided by the application.

The LCD panel is located on the front of the Base Server chassis. The PS-1 and the DEI do not have LCD panels.

LCD Panel and Controls

Figure 4 shows an example of the 2-line LCD panel, the cursors that indicate the current selection and additional options, and the controls used to navigate the LCD application menus and options.

Figure 4: LCD Panel and Controls



The LCD contains two rows for displaying American English text. Each row contains 16 character spaces. If the text is too long for the current line, it is wrapped to the next line. The first and last columns of the two rows are reserved for cursors. The left cursor indicates the current selection. The right up and down cursors indicate additional options. If no additional options are available, the up and down cursors are not displayed.

There are five buttons located next to the display. The four triangular buttons are used to navigate through the menus options that are displayed by moving the cursor up ▲, down ▼, left ◀, and right ▶. The round center button ● is used to select the option currently displayed or confirm your selection. It is also used to power up the server if it has been manually shut down.

Application Software and Security

By default, the system uses DHCP to obtain system network settings. Mitel recommends that you do the following using the LCD application when you complete the initial system configuration:

- Replace the DHCP network settings with static IP settings.
- Disable DHCP.

The LCD application also provides access to other system configuration options and actions and communicates with Call Processing via the Resource Command Handler (RCH).

The LCD application does not require you to enter a password to access any of the system menus or options via the LCD panel and buttons. Therefore, the system should be installed in a secure location. Only certified Mitel technicians should have access to the MiVoice Office 250 hardware and software.

Idle Display

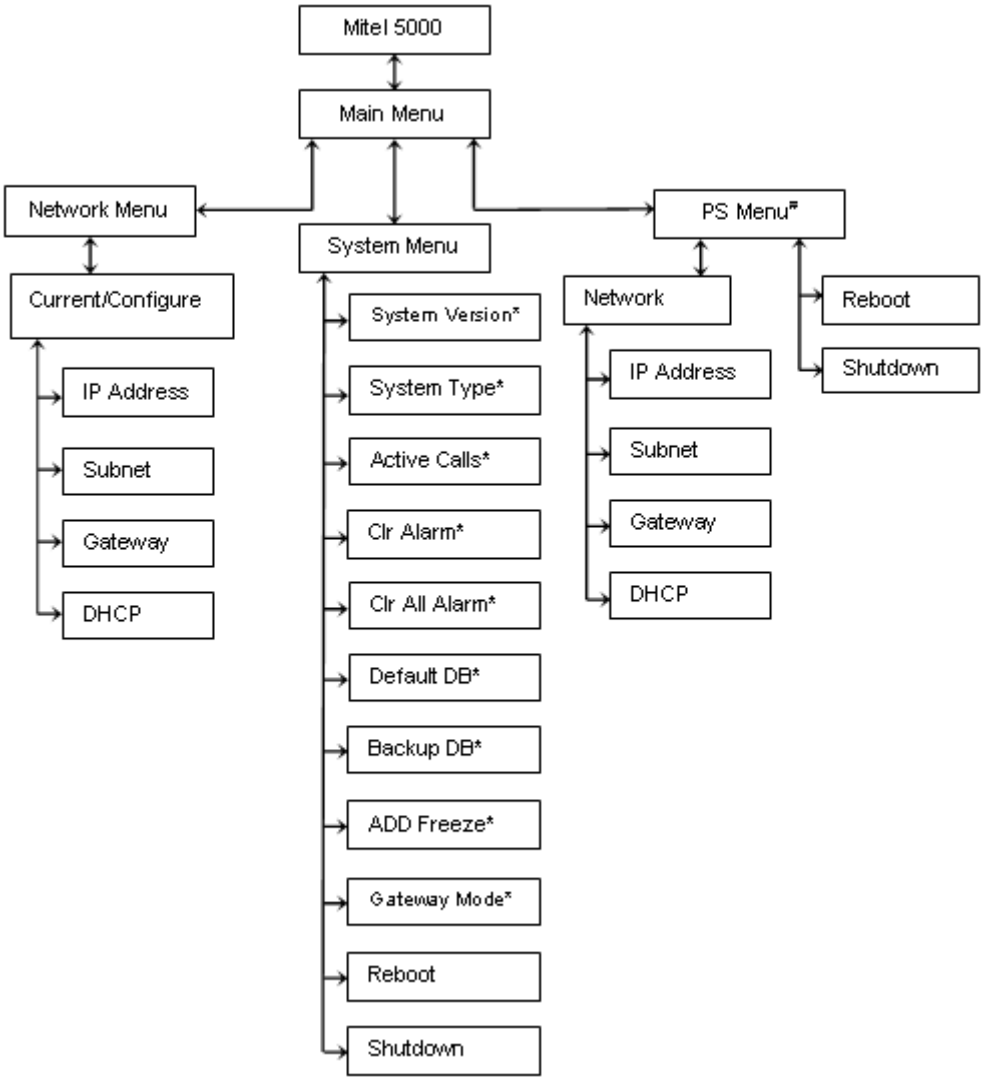
The idle display may include the hostname for the server (**MiVoice Office 250 <*Hostname*>**). The hostname is configurable in Database Programming. If you change the system's hostname, the new hostname is displayed within 15 seconds. If the hostname is blank, the idle display shows only **MiVoice Office 250**.

Menus and Options

The LCD software application provides a way to view and edit network and system settings displayed in the LCD panel. LCD menus time out after approximately 15 seconds of inactivity. When a menu times out, the idle display (**MiVoice Office 250**) appears.

Figure 5 provides a high-level overview of the LCD application's available menus.

Figure 5: LCD Application Menus



* This menu item is not available unless Call Processing is online.
\# The PS menu is not available unless a PS is networked with the Base Server.

To navigate to and select menu options:

1. Press the up or down arrow buttons to see the available options.
2. When the appropriate option is displayed, press the center button to select it.
3. Press the center button again to confirm the selection.

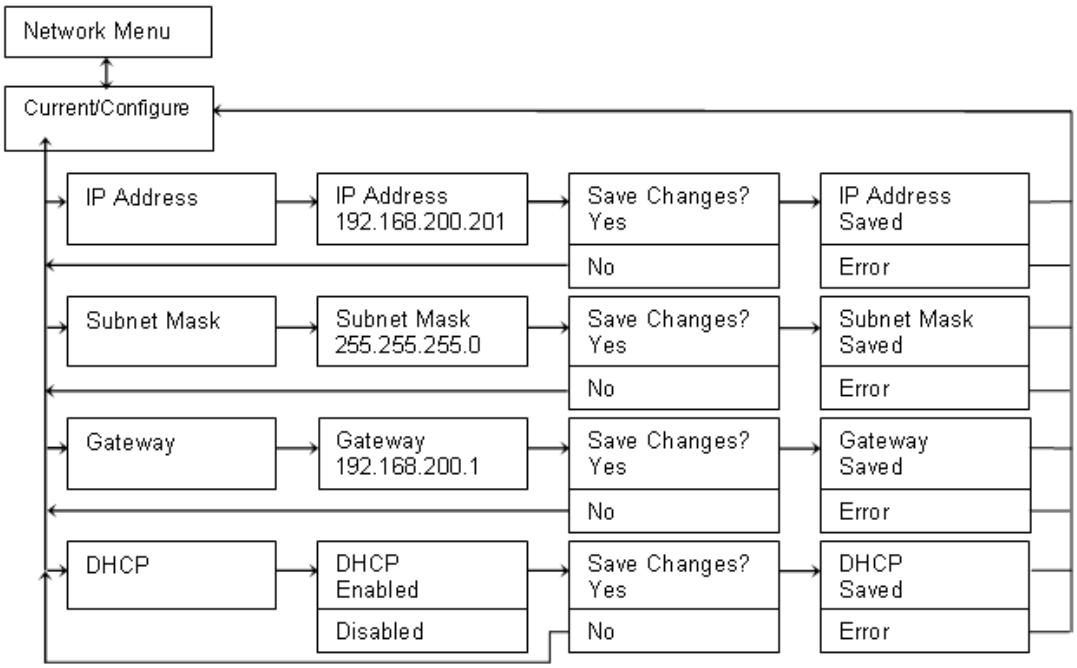
Network Menu

The Network menu provides a way to view and configure the IP address, subnet mask, default gateway, and DHCP options for the server. By default, DHCP is enabled, which provides system network settings when the server is first powered up. However, Mitel recommends that you change the DHCP settings to static IP settings and disable DHCP when you complete the initial configuration of the system. For information about the initial configuration of the system, see “Setting Initial System Configuration“ on **page 166**.

After you configure static IP settings, the new values are sent to Call Processing and saved in the database. When you reboot the server, the new settings take effect. If Call Processing is not running when you change the network settings, the LCD application stores this information until Call Processing is started and you reboot the server.

To view/edit Network menu settings:

1. From the idle menu, press the down arrow button to navigate to the Network menu.



2. Press the down arrow button to view/configure the following:
   - **IP Address**: Use the static IP settings provided by your network administrator.
   - **Subnet Mask**: Use the static IP settings provided by your network administrator.
   - **Gateway**: Use the static IP settings provided by your network administrator.
   - **DHCP**: Disable DHCP.

   To edit an IP address:

   a. Press the left arrow when the cursor is blinking on the left digit (of the address you want to edit). This deletes the address.

   b. Use the up and down arrow buttons to change the digits.

c. Press the center button to save your selection. The **Save Changes?** prompt appears.

d. Press the center button again to confirm the selection.

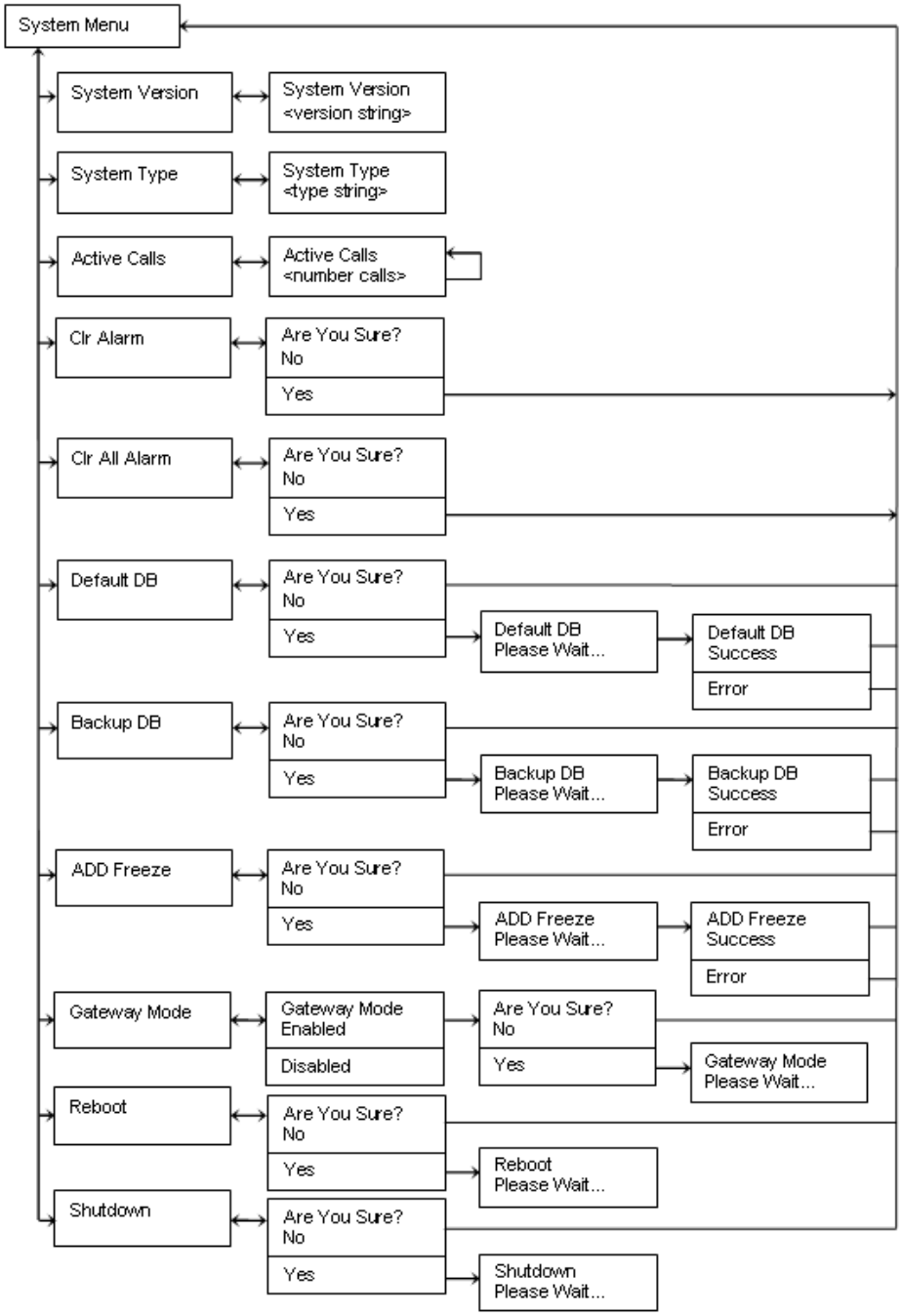
3. After you change the system's network settings, reboot the system from the System menu (see page 175).

System Menu

The System menu includes options to view system settings and perform several actions. Call Processing must be online for all of the System menu options except Reboot and Shut down.

To view/configure System menu settings:

1. From the idle menu, press the down arrow button to navigate to the System menu.



2. Press the down arrow button to view/configure the following:
   - **System Version**: Displays the Call Processing software version, which is used as an overall version for the system.
   - **System Type**: Displays the current system type as specified by the Call Processing software.

- **Active Calls**: Displays the total number of active calls including phone calls, trunk calls, voice mail calls, faxes etc. When you are viewing the active calls, you can refresh the screen by pressing any button except the left arrow button.
- **Clear Alarm**: Clears the alarm message that displaced the idle menu.
- **Clear All Alarms**: Clears all system alarms.
- **Default Database**: Sends a command to Call Processing to default the database. After the database is defaulted, the system automatically reboots. Note that this action will drop all current calls.
- **Backup Database**: Sends a command to Call Processing to perform a database backup. If the backup fails, the LCD displays **Error**.
- **ADD Freeze**: Sends a command to Call Processing to perform an ADD freeze operation. If the operation fails, the LCD displays **Error**.
- **Gateway Mode**: Sends a command to Call Processing to configure the server as a Base Server for use with a Processing Server. Call Processing performs a reset to complete the operation. If the operation fails, the LCD displays **Error**.
- **Reboot**: Sends a reset command to Call Processing. Note that this action will drop all current calls.
- **Shut Down**: Sends a shut down command to Call Processing. Note that this action will drop all current calls. After a shut down you can restart the server manually by pressing the center button.

  When prompted to confirm your action, press the down arrow button to select **Yes**, or the up arrow button to select **No**. Wait until the action is completed.

To complete an ADD freeze from the System Menu:

1. From the idle menu (**MiVoice Office 250**), press the down arrow button to navigate to the System menu, and then the ADD Freeze option.
2. Press the center button to select the **ADD Freeze** option. The **Are You Sure?** prompt appears.
3. Press the up arrow button to select **Yes**, and then press the center button to confirm your selection. The ADD freeze request is sent to Call Processing and the **Please Wait** message is displayed. Once the request has been sent the **Success** or **Error** message is displayed.
   - A **Success** message indicates that the request was sent successfully.
   - An **Error** message indicates that ADD has not been configured or enabled in DB Programming, or that a freeze is already in progress.

   After approximately 10 seconds, you are returned to the System menu. The LCD Application does not wait for the freeze operation to complete.

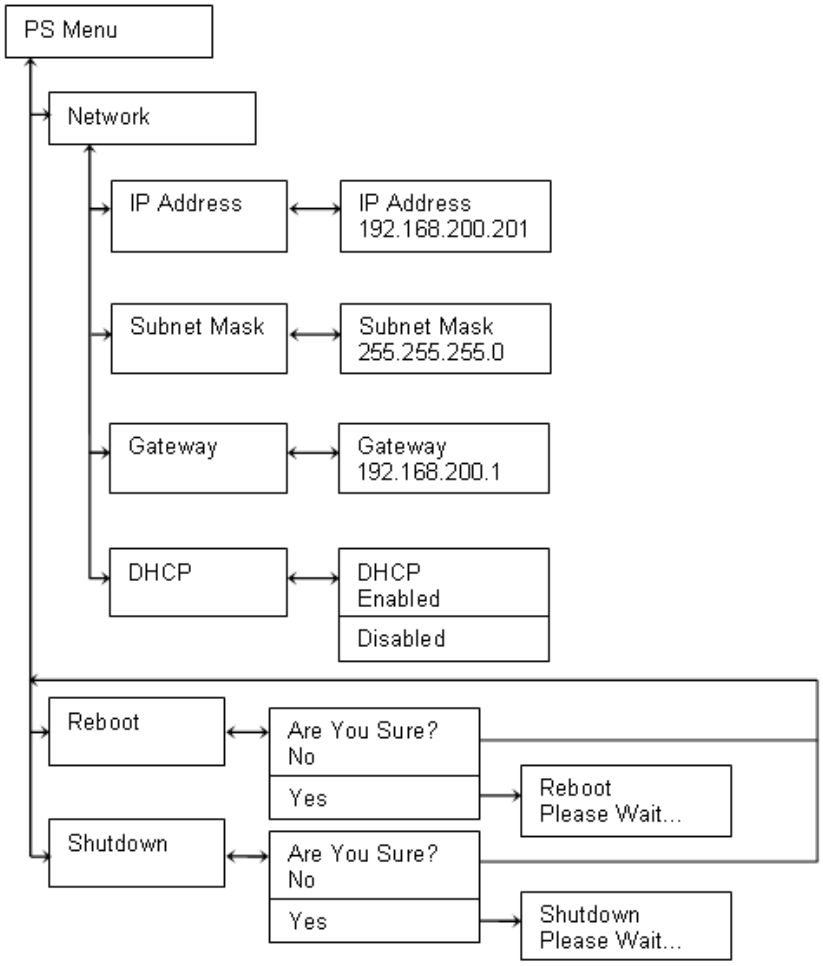
NOTE If you select **No**, the ADD freeze is not completed and the LCD application returns to the System menu.

Processing Server (PS) Menu

From the PS menu you can view (but not configure) the IP address, subnet mask, default gateway, and DHCP setting for a connected PS-1. You can also reboot and shut down the PS-1 from the PS menu. The PS menu is only available if a PS-1 is connected to the Base Server, and the server is online.

To view/configure PS menu settings:

1. From the idle menu, press the down arrow button to navigate to the PS menu.



2. Press the down arrow button to view the following:
   - IP Address
   - Subnet Mask
   - Gateway
   - DHCP
3. You can perform the following actions from the PS menu:
   - **Reboot**: Reboots the PS and all connected servers. Note that this action will drop all current calls.

- **Shut Down**: Shuts down the PS an all connected servers. Note that this action will drop all current calls. After a shut down you can restart the server manually by pressing the center button.

  When prompted to confirm your action, press the down arrow button to select **Yes**, or the up arrow button to select **No**. Wait until the action is completed.

Additional System Messages

The following system messages are also supported by the LCD application:

- **Alarm messages**: When an alarm is generated, the alarm message displaces the idle display on the LCD. Alarm messages are displayed on the LCD, one at a time, similar to the way they are displayed on an administrator phone. The System menu provides a options to clear a single alarm or clear all alarms (see page 175).
- **NOR flash and Field Programmable Gate Array (FPGA) Updates**: When you first power up the server, the system checks to see if any NOR flash or FPGA components require an update. A series of messages, displayed on the LCD, prompt you to perform the required updates. During a NOR flash or FPGA update, it is critical that the server remain up and running while the updates are being completed.

Troubleshooting

If an **Error** message appears on the LCD after you attempt to make a system change, examine the Resource Command Handler (RCH) log files via the Administrative Web Portal (AWP) pages, to determine the cause of the error. You may need to adjust the level of logging and/or reproduce the error message to view it in the log files.

The Log Files page is available for MiVoice Office 250 systems, however, the RCH log file applies to Base Servers only.

To view the RCH log files:

1. Open a Web browser and navigate to the MiVoice Office 250 Administrative Web Portal pages. The Enter Network Password dialog box appears.
2. Type your User Name and Password in the text boxes and click **OK**. The Home page appears.
3. *If you are currently viewing the Web pages for the Processing Server*, click **Switch to Base Server.**
4. Click the **Logging** navigation tab. The Log Files page appears.
   Look under the *System Resource Application* heading.
5. Click the link to the RCH log file to open it and view the contents. Messages from the LCD application begin with **LCD APP**.

Flash Memory and Low-Level Firmware Update Messages

For v3.0 and later, new LCD messages, designed to safeguard low-level firmware updates, have been added to the LCD application.

This update is required when you:

- perform the initial configuration of the system.
- install a new compact flash-type memory card.
- swap a processor module.
- perform a system upgrade.

During this update, it is critical that the server remain up and running while the updates are being completed. Do not power off the server during an update. If any of the updates are missing or become corrupted, the server will not operate and must be returned to Mitel.

CAUTION

Potential system failure. Do ***not*** shut down, reboot, or remove power from the server while the components are being updated. Shutting down, rebooting, or removing power from the server before the update is complete will render the server ***inoperable*** and you will need to return the server to Mitel.

Flash Memory Updates

The MiVoice Office 250 server processor modules have 4 MB of flash memory. When you first power up the server, the system software detects if any flash memory components require an update. If an update is required, the following message appears on the LCD:

“Upgrade Norflash enter any key.”

Press any button next to the LCD panel to acknowledge the message, and start the update.

NOTE When you perform a system upgrade using the AWP Web pages, flash memory updates are completed automatically.

During the update process, the following messages alternately appear on the LCD:

“Upgrading.....<*NOR flash component*>.”

“Do not power off.”

Each component can take several seconds to update. ***Do not shut down, reboot, or power down the server during the update process.***

After the components are updated, the system automatically reboots.

Low-level Firmware Updates

The MiVoice Office 250 motherboard is equipped with low-level firmware that controls electrical signals to the hardware devices such as compact flash-type memory cards, Universal Serial Bus (USB), Digital Expansion Interfaces (DEIs), etc.

After flash memory updates, the system software detects if low-level firmware requires an update. If an update is required, the following message appears on the LCD:

“Upgrade FPGA enter any key.”

Press any button next to the LCD panel to acknowledge the message, and start the update.

NOTE When you perform a system upgrade using the AWP Web pages, low-level firmware updates are completed automatically.

During the update process, the following messages are alternately appear on the LCD:

“Upgrading.....FPGA.”

“Do not power off.”

The update requires approximately 30 seconds. ***Do not shut down, reboot, or power down the server during the update process.*** After the update is complete, the system automatically reboots.

When all of the flash memory and low-level firmware components have been updated, the LCD displays the **MiVoice Office 250** idle screen.

BASE SERVER MODULES

This section provides procedures for installing and removing the expansion modules that can be installed in the MiVoice Office 250 Base Server. The procedure for installing a PEC-1 on an HX Controller PM or a CS Controller PM-1 also appears in this section.

With the exception of the HX Controller PM or CS Controller PM-1, the other expansion modules can be installed in any bay. However, if a module was programmed while in one bay, it will need to be reprogrammed if moved to another bay.

NOTICE

Electrostatic Discharge (ESD). The modules and components are static-sensitive. Handle the modules by the edges only and keep them in their protective bags until they are to be installed. Do not bend or touch the component pins or subject them to a static discharge. When working with the modules, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the module during shipping. Any static charge (no matter how small) must be discharged from the body before touching the modules or other components. *The warranty does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will **not** be replaced under warranty.*

Installing an HX Processor Module (PM)

Prior to mounting the HX Controller chassis, you must first insert the HX Processor Module in the chassis as described in the following procedure. For associated part numbers, see page 371.

NOTICE

Equipment Damage Hazard. Inserting or removing a processor module with AC power turned on to the chassis will damage the processor module. A processor module damaged in this manner will ***not*** be replaced under warranty.

Electrostatic Discharge (ESD). The modules and components are static-sensitive. When working with the modules, use a properly terminated anti-static wrist strap. Any static charge (no matter how small) must be discharged from the body before touching the modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will not be replaced.*

To install the HX Processor Module:

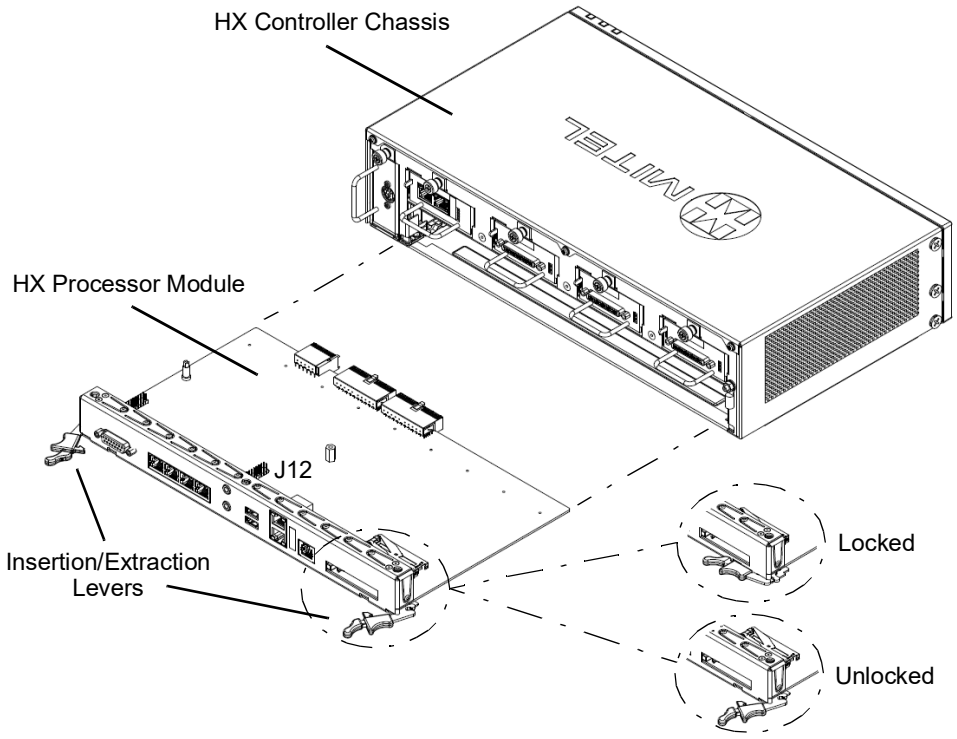
1. Make sure the HX Controller chassis is not connected to a power source. See the NOTICE above.
2. Using a properly terminated anti-static wrist strap, remove the HX Processor Module from its protective anti-static bag.
3. Using Figure 5 on **page 181** as a guide, check to make sure that a block strap is installed over the first seven pair of pins on the mount point closest to the center of the module (connector J12).
4. If a Processor Expansion Card (PEC-1) is required, follow the instructions for “Adding a Processor Expansion Card (PEC-1)“ on **page 181** prior to inserting the HX Processor Module into the HX Controller chassis.
5. Make sure the two insertion/extraction levers located at the far left and far right sides of the HX Processor Module are in the unlocked position (as shown in **Figure 5**).

6. Carefully insert the HX Processor Module into the large processor module bay in the HX Controller chassis (the bottom bay looking at the back of the chassis). Make sure the module aligns with the rails inside the chassis, then carefully slide the module into the chassis until it touches the back of the chassis.
7. Push the two insertion/extraction levers in towards each other until they lock the module into place.

NOTICE

Equipment Damage Hazard. Damage to circuit boards may occur if both insertion/extraction levers are not activated at the same time.

Figure 5: HX Processor Module Installation



NOTE To eject the HX Processor Module from its bay and remove it from the chassis (if necessary), follow the procedure outlined in **step 1** in the following section.

Adding a Processor Expansion Card (PEC-1)

If necessary, an optional Processor Expansion Card (PEC-1) can be added to the HX Processor Module to provide additional IP and/or voice processing resources. See the *MiVoice Office 250 Engineering Guidelines* for details on when a PEC-1 might be needed.

To install the Processor Expansion Card:

1. If performing this procedure on a system that has already been installed, follow the substeps below to remove the HX Processor Module. If not, proceed to **step 2**.

a. Perform a system and voice mail database backup using DB Programming.

b. Shut down the system using the LCD panel. Unplug the external power supply unit from the AC power source.

c. Wearing a properly terminated anti-static wrist strap, pinch the two spring-loaded handles on the insertion/extraction levers located at the far left and far right sides of the HX Processor Module, and then push the levers out away from each other to unlock and eject the module from its bay (as shown in **Figure 5** above).

d. Carefully slide the module out from the chassis.

2. Using **Figure 3** below as a guide, do the following:

a. Place the processor module on a clean, ESD-protective surface

b. On the processor module, remove the strap across the first seven pair of pins on the mount point closest to the center of the module (connector J12). Do not discard the strap.

IMPORTANT	The pin strap on the mount point (J12) must be removed before installing the PEC-1. If the PEC-1 is later removed, the strap MUST be re-installed, or the processor module will not function.

c. Check the standoffs (supports). For guidance, see **Figure 3**.

- **Hexagon-shaped metal standoff**: Check that the metal standoff is in place on the processor module. It is located toward the middle of the processor module.
- **Plastic standoff**: Check that the thicker end of the standoff is securely pressed into the hole on the processor module.

3. Mount the PEC-1 onto the processor module. See **Figure 3** as a guide.

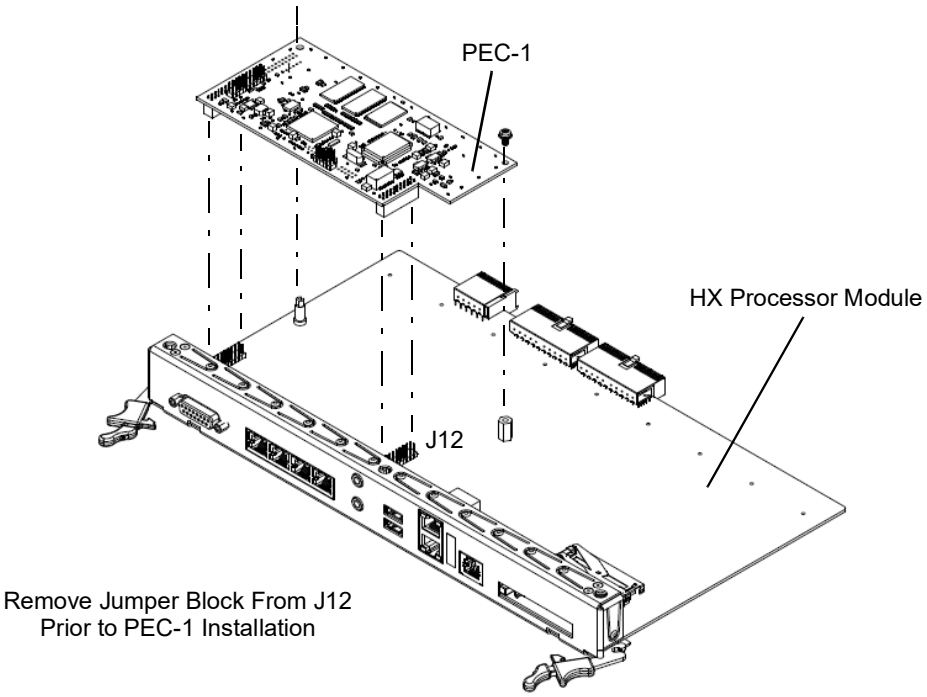
a. Align the PEC-1 mount points/holes over the mount points and stand-offs on the processor module.

b. Carefully press the PEC-1 into place on the processor module. Be careful not to bend any pins or force the card onto the module.

4. Secure the PEC-1 to the processor module using the supplied screw (located in a small plastic bag attached to the PEC-1 anti-static bag). This screw attaches to the hexagon-shaped metal stand-off on the processor module. Do *not* overtighten this screw.

5. Insert the processor module into the HX Controller chassis as outlined beginning in step 5 on **page 180**.

Figure 3: Processor Expansion Card (PEC-1) Installation



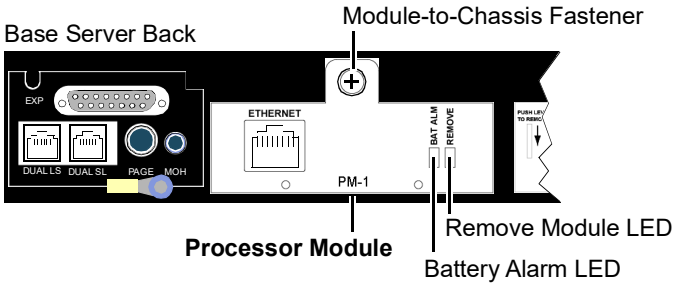
Installing a CS Controller Processor Module (PM-1)

Install the Processor Module (PM-1) in the CS Controller chassis as described in the following procedure. For part numbers see Table 59, “Replacement Parts,” on **page 372**. **Figure 4** provides an illustration of the PM-1.

NOTICE

Equipment Damage Hazard. Use the LCD panel to shut down the system, then unplug all power sources to the chassis before working with the PM-1. Replacing or inserting a PM-1 with AC power turned on to the chassis will damage the module. A PM-1 damaged in this manner will ***not*** be replaced under warranty.

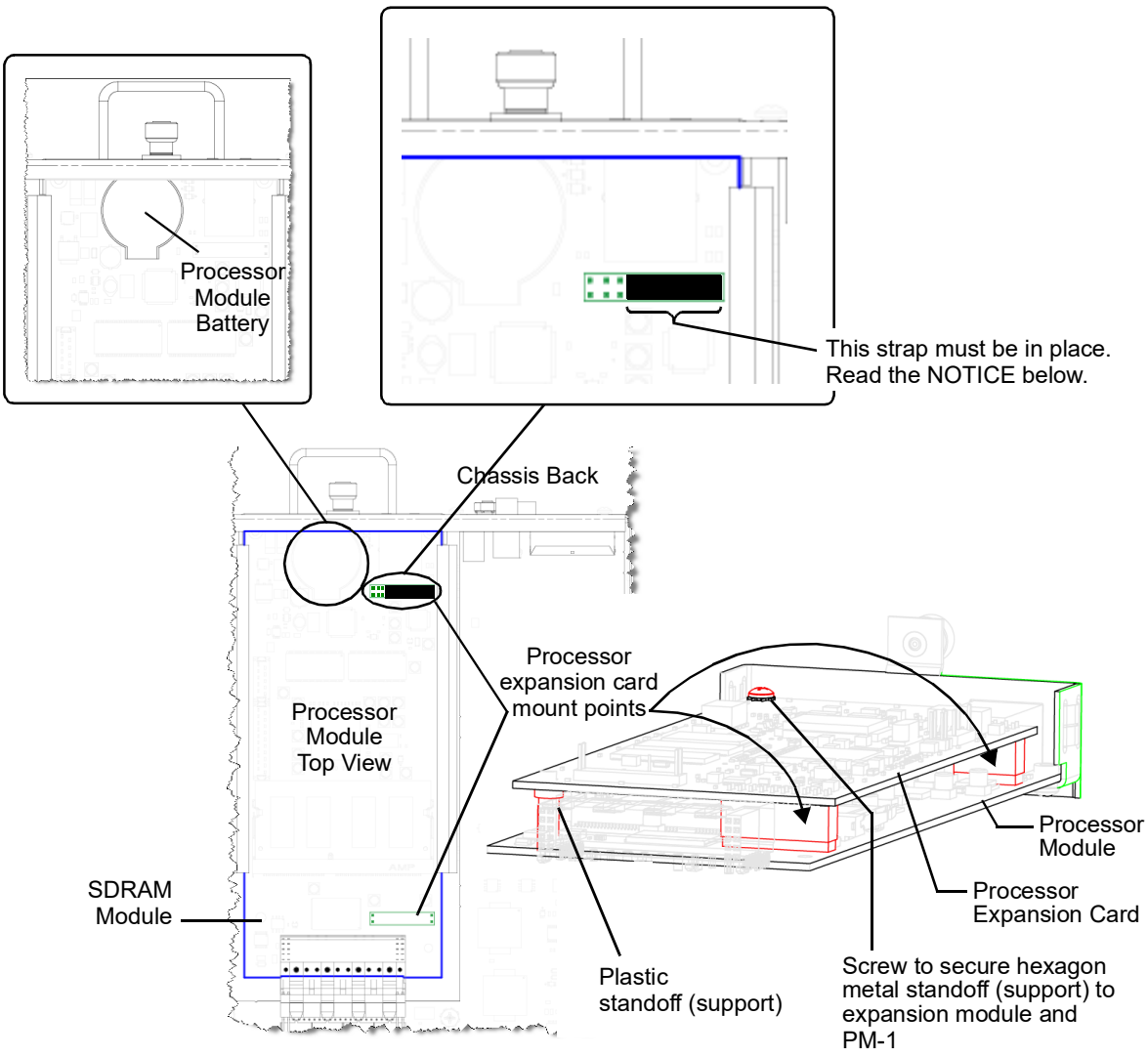
Figure 4: Processor Module Bay Location



To install the PM-1:

1. Make sure the system has been shut down (using the LCD panel) and that the power cord has been unplugged from the power source. See the NOTICE above.
2. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag.
3. Check the following components. Use the following illustration as a guide.
   - **Module battery**: Check that the PM-1 battery is securely in place.
   - **SDRAM memory**: Check that the SDRAM memory is correctly seated (the module is pushed in and the gold “fingers” are not exposed).
   - **Pins 1–7 block strap**: Check that the strap is installed on pins 1 through 7.

Figure 5: Processor Module Components



NOTICE

System Inoperability. A strap covers pin 1 through pin 7 (far right in the illustration above. This strap must be in place, unless using the PEC-1 (page 186). If a system ever needs to function without the PEC-1 (for example, replacing the expansion card), the strap will need to be put back on the PM-1. The standard PM-1 (without a PEC-1) will not function without the strap in place.

4. Insert the module carefully into the first bay (left side looking at the back of the chassis). Make sure the module aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops. Do not use force.
5. Tighten the fastener at the top of the module, securing it to the chassis. Do *not* over tighten this fastener.

6. If not adding any other modules, continue to “Installing the USB Security Key and Compact Flash-Type Memory Card” on page 150. Otherwise, continue to “Installing a Two-Port Loop Start Module (LSM-2)” on **page 1-189**.

To remove a PM-1:

1. Use the LCD panel on the front of the chassis to shut down the system.
2. Unplug the chassis power cord from the power source.
3. Loosen the fastener at the top of the module (this fastener secures the module to the chassis).
4. Using a properly terminated, anti-static wrist strap, remove the module from the chassis and place it into the anti-static bag that came with the module.

Adding a Processor Expansion Card (PEC-1)

An optional Processor Expansion Card (PEC-1) provides additional IP resources. See the **Specifications** and **Product Description** chapters for PEC-1 details. The PEC-1 includes the necessary hardware to mount it to the PM-1.

To install the PEC-1 on a CS Controller PM-1:

1. Perform a Database backup using DB Programming. You can also back up the database using the LCD panel. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.
2. Perform a voice mail backup using a USB flash drive attached to the USB-A port on the front of the CS Controller. Do not back up VM to the compact flash-type memory card. Refer to the *MiVoice Office 250 Features and Programming Guide*, for details on backup up either Unified Voice Messaging or external voice mail.
3. Shut down the system using the LCD panel. See Figure 4 on **page 169** for an illustration of the menu locations. Unplug the chassis power cord.

NOTICE

Equipment Damage Hazard. Use the LCD panel to shut down the system, and unplug all power sources to the chassis before working with the PM-1. Replacing or inserting a PM-1 with AC power turned on to the chassis will damage the PM-1. A PM-1 damaged in this manner will not be replaced under warranty.

Electrostatic Discharge (ESD). The modules and components are static-sensitive. When working with the modules, use a properly terminated anti-static wrist strap. Any static charge (no matter how small) must be discharged from the body before touching the modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will not be replaced.*

4. Wearing a properly terminated anti-static wrist strap, loosen the fastener at the top of the PM-1. Remove the PM-1 from the chassis and place it on a clean, ESD-protective, surface.
5. Using Figure 3 on **page 188** as a guide, do the following:
   a. On the PM-1, remove the strap across the first seven pins on the mount point closest to the front of the chassis. Do not discard the strap. Please read the important message below.

IMPORTANT	The pin strap on the front expansion card mount point must be removed before installing the PEC. Also, if the PEC is removed and/or the PM-1 returned, the strap MUST be in place on the first seven pairs of pins, or the PM-1 will not function.

b. Check that the SDRAM memory is correctly seated. Visually inspect to make sure the module is pushed in and the gold “fingers” are not exposed.

c. Check the standoffs (supports). For guidance, see **Figure 3**.

- **Hexagon-shaped metal standoff**: Check that the metal standoff is in place on the PEC-1. It is located toward the front of the module faceplate.
- **Plastic standoff**: Check that the thicker end of the standoff is securely pressed into the hole on the PEC-1.

6. Mount the PEC-1 onto the PM-1. See **Figure 3** as a guide.

a. Carefully align the PEC-1 mount points and the stand-off mount points.

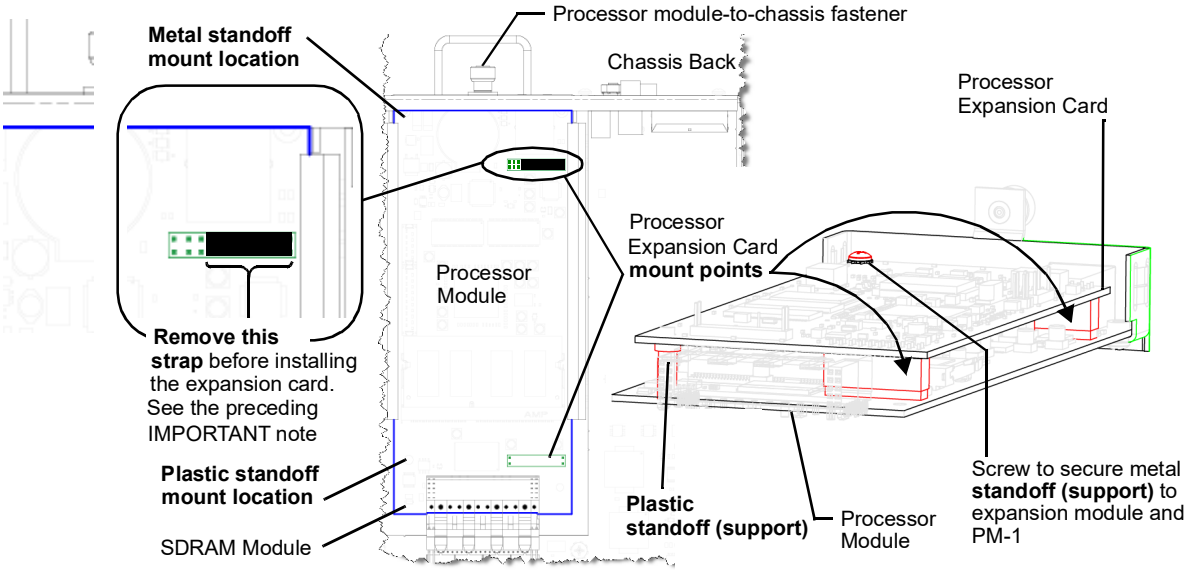
b. Gently press the smaller end of the standoff onto the PM-1 while aligning the screw mount point over the hexagon-shaped metal standoff and the PEC mount points. Do not force the card onto the module.

7. Secure the PEC-1 to the PM-1 using the supplied screw (located in a small plastic bag attached to the PEC-1 anti-static bag). This screw attaches to the hexagon-shaped metal stand-off installed in **step 5**c on **page 1-187**. Do *not* overtighten this screw. Insert the PM-1 into the chassis as described in the procedure “With the exception of the HX Controller PM or CS Controller PM-1, the other expansion modules can be installed in any bay. However, if a module was programmed while in one bay, it will need to be reprogrammed if moved to another bay.” on **page 1-180**.

8. Check that all cables are connected and plug in the AC power cord. The system boots and when finished, it displays MiVoice Office 250 on the LCD panel. This indicates the system is ready to be programmed.

9. Refer to the *MiVoice Office 250 Features and Programming Guide*, for information to load the license file. You can then program the additional IP resources.

Figure 3: Expansion Card Installation



Installing a Two-Port Loop Start Module (LSM-2)

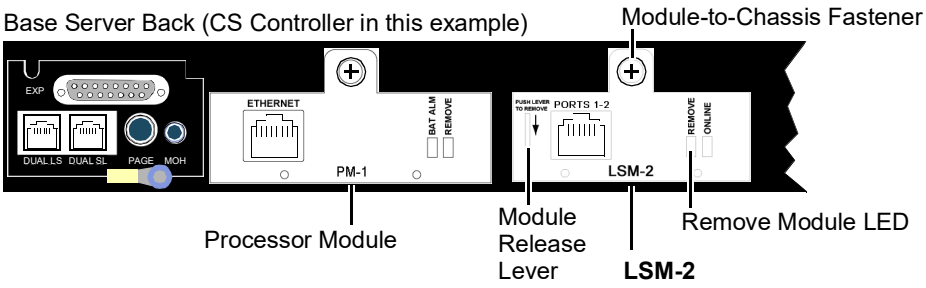
Install the two-port loop start module (LSM-2) as described below. Part numbers appear in Table 59, “Replacement Parts,” on **page 372**. See Figure 4 on **page 189**.

IMPORTANT	The loop start module can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever down until the LED indicator turns green allows the software to “disconnect” from the module. Failure to wait until the LED turns green, could result in pending information getting lost or unexpected system behavior. Other than the PM-1, all other modules can reside in any bay. However, DB Programming will need to change, if you move a module (that was previously programmed) into a different bay location.

To install the two-port Loop Start module (LSM-2):

1. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag.
2. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops and the module release lever clicks into place (up position). Do not use force.
3. Tighten the fastener at the top of the module, securing it to the chassis. Do not over tighten this fastener.
4. If not adding any other modules, continue to “Installing the USB Security Key and Compact Flash-Type Memory Card” on page 150. Otherwise, continue to “Installing a Single-Port T1/E1/PRI Module (T1M)” on page 194.

Figure 4: LSM-2 Module



To remove the two-port Loop Start module (LSM-2):

1. Using a properly terminated, anti-static wrist strap, loosen the fastener at the top of the module. This fastener secures the module to the chassis.
2. Push down and hold the module release lever.
3. When the REMOVE LED turns green, grasp the module handle and carefully remove the module from the chassis.

Installing a Four-Port Loop Start Module (LSM-4)

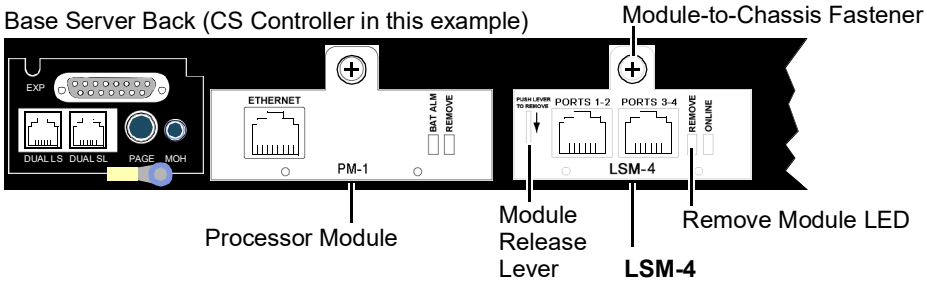
The LSM-4 provides four trunk interface ports. For additional technical data about the LSM-4, see the **Specifications** chapter. See page 154 for information on how this module is cabled to the MDF.

IMPORTANT	The LSM-4 can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever until the LED indicator turns green allows the software to "disconnect" from the module. Failure to wait until the LED turns green, could result in pending information getting lost or unexpected system behavior. LSM-4 modules can reside in any bay. However, DB Programming will need to change, if you move a module (that was previously programmed) into a different bay location.

To install an LSM-4:

1. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag. For an example of an LSM-4 module that you can see for the remaining steps in this procedure, see **Figure 5**.
2. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops. Do not use force.
3. Tighten the fastener at the top of the module, securing the module to the chassis. Do not over tighten this fastener.
4. Continue to "Verifying and Securing Cable Connections" on page 165.

Figure 5: LSM-4 Module



To remove an LSM-4:

1. Using a properly terminated, anti-static wrist strap, loosen the fastener at the top of the module (this fastener secures the module to the chassis).
2. Push down and hold the module release lever.
3. When the **REMOVE** LED turns green, grasp the module handle and carefully remove the module from the chassis.

Installing a Dual T1/E1/PRI Module (T1M-2)

This procedure assumes the required pre-installation checklist items, regarding T1/E1/PRI, are in place. If necessary, see “Gathering the Necessary Tools and Supplies“ on **page 134**.

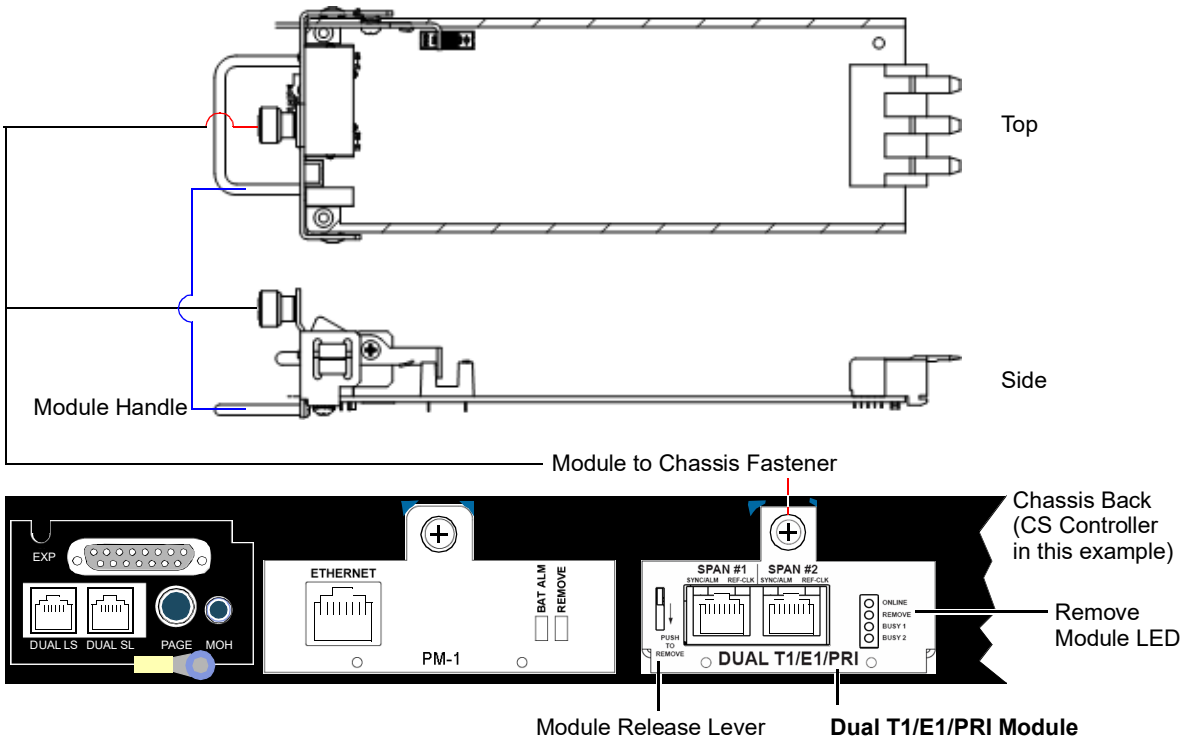
The Dual T1/E1/PRI module is installed and configured like the single T1/E1/PRI module with the following exception: Each port on the dual module can be configured differently from the other. For example, you can configure one port for T1 and the other for T1/PRI or E1/PRI. The single T1 module provides two ports as well, but one of the ports is for testing purposes. Unlike the single T1/E1/PRI module, the dual module does not provide the test port.

IMPORTANT	The Dual T1/E1/PRI module can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever until the LED indicator turns green allows the software to “disconnect” from the module. Not waiting until the LED turns green could result in pending information getting lost or unexpected system behavior. Other than the PM-1, all other modules can reside in any bay. However, DB Programming will need to change, if you move a module (that was previously programmed) into a different bay location.

To install a Dual T1/E1/PRI module:

1. Using a properly terminated anti-static wrist strap, remove the module from the protective anti-static bag.
2. Insert the module carefully so that it aligns with the rails inside the chassis, and then gently slide the module into the chassis until it stops and the module release lever clicks into place (up position). Do not use force. See **Figure 6** below for an illustration of this module.
3. Tighten the fastener at the top of the module, securing the module to the chassis. Do not over tighten this fastener.

Figure 6: Dual T1/E1/PRI Module



To remove a dual T1/E1/PRI module:

1. Using a properly terminated anti-static wrist strap, loosen the fastener at the top of the module (this fastener secures the module to the chassis).
2. Push down and hold the module release lever, and then when the **REMOVE** LED turns green, grasp the module handle and carefully remove the module from the chassis.

Dual T1/E1/PRI Module LED Summary

The Dual T1/E1/PRI module has a total of 8 LEDs on the faceplate to indicate module status. The LEDs are contained in two separate devices: a 4-LED stack, and 4 bi-color LEDs integrated into the RJ-45 connector.

The LED stack has 2 green LEDs and 2 yellow LEDs. These LEDs are described in **Figure 7** and in tables **Table 31** and **Table 32** on page 193.

Figure 7: Dual T1/E1/PRI Module Faceplate LED Summary

Table 31: Dual T1/E1/PRI Module LED Summary — 4-LED Stack

LED Name	Color	Indication
Online	Green #1	ON: Board is online and functioning normally. OFF: Board is not online.
Remove	Green #2	ON: Board is safe to remove. OFF: Board is not ready to be removed
Busy 1	Yellow #1	ON: One or more channels on the first span are active. OFF: No channels on the first span are active.
Busy 2	Yellow #2	ON: One or more channels on the second span are active. OFF: No channels on the second span are active.

Table 32: Dual T1/E1/PRI Module LED Summary — RJ-45 LEDs

LED Name	LED	Color	Indication
SYNC/ALRM Span 1 (synchronization and alarm status)	1	Green or Red	Solid green: Stable state. Blinking green: Initial trouble state or clearing trouble state. Solid red: Stable error state. Blinking red: Clearing trouble state. OFF: Span is disabled / not active.
REF_CLK Span 1 (reference clock)	2		Green ON: Span provides the reference clock for the system. OFF: Span does not provide the reference clock for the system.
SYNC/ALRM Span 2 (synchronization and alarm status)	3		Solid green: Stable state. Blinking green: Initial trouble state. Solid red: Stable error state. Blinking red: Clearing trouble state. OFF: Span is disabled / not active.
REF_CLK Span 2 (reference clock)	4		Green ON: Span provides the reference clock for the system. OFF: Span does not provide the reference clock for the system.

Installing a Single-Port T1/E1/PRI Module (T1M)

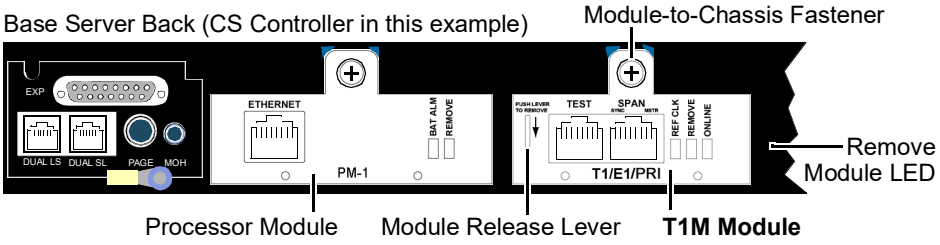
See Figure 8 on **page 194** for an illustration of this module and for the following procedure.

IMPORTANT	The T1M module can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever until the LED indicator turns green allows the software to "disconnect" from the module. Failure to wait until the LED turns green, could result in pending information getting lost or unexpected system behavior. Other than the PM-1, all other modules can reside in any bay. However, DB Programming will need to change, if you move a module (that was previously programmed) into a different bay location.

To install a T1M module:

1. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag.
2. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops and the module release lever clicks into place (up position). Do not use force.
3. Tighten the fastener at the top of the module, securing the module to the chassis. Do not over tighten this fastener.
4. If not adding any other modules, continue to "Installing the USB Security Key and Compact Flash-Type Memory Card" on page 150.

Figure 8: T1M Module



To remove a T1M module:

1. Using a properly terminated, anti-static wrist strap, loosen the fastener at the top of the module (this fastener secures the module to the chassis).
2. Push down and hold the module release lever, then when the **REMOVE** LED turns green, grasp the module handle and carefully remove the module from the chassis.

Single T1/E1/PRI Module LED Summary

The single T1/E1/PRI module has a total of 7 LEDs (two unused) on the faceplate to indicate module status. The LEDs are contained in two separate locations: 3 LEDs to the right side of the module faceplate, and 2 green LEDs integrated into each RJ-45 connector.

These LEDs are described in **Figure 9** and in tables **Table 33** and **Table 34** below.

Figure 9: Single T1/E1/PRI Module Faceplate LED Summary

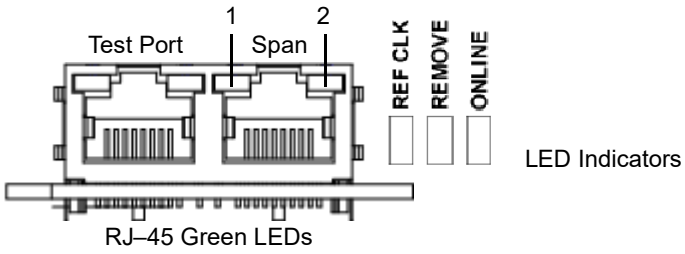


Table 33: Single T1/E1/PRI Module LED Summary — 3 LEDs

LED Name	Color	Indication
REF CLK	Green	ON: Span provides the reference clock for the system. OFF: Span does not provide the reference clock for the system.
REMOVE		ON: Board is safe to remove. OFF: Board is not ready to be removed
ONLINE		ON: Board is online and functioning normally. OFF: Board is not online.

Table 34: Single T1/E1/PRI Module LED Summary — RJ-45 LEDs

LED Name	LED	Color	Indication
Test port LEDs	N/A		Not used
SYNC (synchronization and alarm status)	1	Green	Solid green: Stable state. Blinking green: Initial trouble state or clearing trouble state. OFF: Stable error state or span is disabled/not active.
MSTR (reference clock)	2		Green ON: Follows the span configuration as programmed in the database. The LED is on if the port/span is programmed for “Master for Private Network,” as selected in the Reference Clock List programming. OFF: Span does not provide the reference clock for the system.

Installing a Basic Rate Module (BRM-2)

The Basic Rate Module (BRM-2) is used for connecting ISDN S-interface trunking. For additional information about modules, see the **Specifications** and **Product Description** chapters.

IMPORTANT

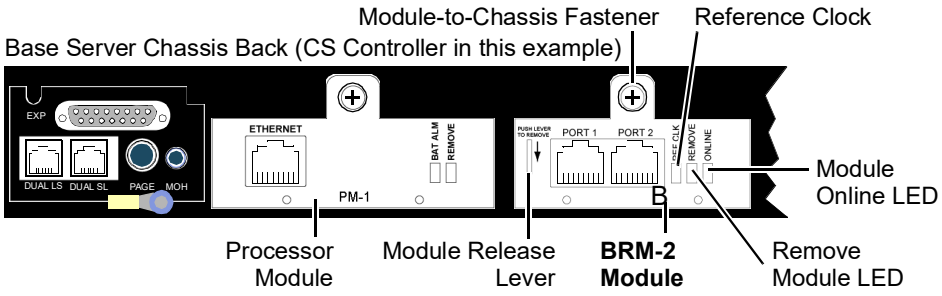
The BRM-2 module can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever until the LED indicator turns green allows the software to "disconnect" from the module. Failure to wait until the LED turns green could result in pending information getting lost or unexpected system behavior.

BRM-2 modules can reside in any expansion bay. However, DB Programming will need to be changed if you move a module that was previously programmed into a different bay location.

To install a BRM-2:

1. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag. See **Figure 10** for an example of the BRM-2 module for the remaining steps in this procedure.
2. If applicable, remove any protective covers over the module bay where you want to install the BRM-2. These covers prevent debris or other foreign matter from entering the system and possibly harming the internal components. These covers should remain in place until you are ready to install a module.
3. Insert the BRM-2 carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops. Do not use force.
4. Tighten the fastener at the top of the module, securing it to the chassis. Do not over tighten this fastener.
5. Continue to "Verifying and Securing Cable Connections" on page 165.

Figure 10: BRM-2 Module



To remove a BRM-2:

1. Using a properly terminated, anti-static wrist strap, loosen the fastener at the top of the module (this fastener secures the module to the chassis).
2. Push down and hold the module release lever, then when the **REMOVE** LED turns green, grasp the module handle and carefully remove the module from the chassis.

BRM-2 Online/Off-line Handling and LED Descriptions

The system recognizes the BRM-2 as online when inserted into a bay previously programmed as a Basic Rate Module. The system auto-equips this module even if it has not been previously programmed in DB Programming, assuming that DB Programming is not in session. However, you must program the ports/circuits for the ports and circuits to come online. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.

The system recognizes the BRM-2 as off-line when removed from the expansion bay. All the ports and circuits are recognized as being off-line and all calls dropped, when the module is removed.

Three LEDs are seen on the BRM-2 module:

- **ONLINE**: (Green) The ONLINE LED is turned ON whenever the module is inserted into the expansion bay and the module is operational. If the bay in which this module is installed had been previously programmed with another module type, then this LED will not light.
- **REMOVE**: (Green) The REMOVE LED is turned ON whenever the module is safe to be removed from the expansion bay. See page 186 for information on removing the module from the chassis.
- **REF CLK**: (Green) The REF CLK LED is turned ON whenever the module has been selected to provide network clock synchronization to the system.

Installing a Digital Desktop Module (DDM-16)

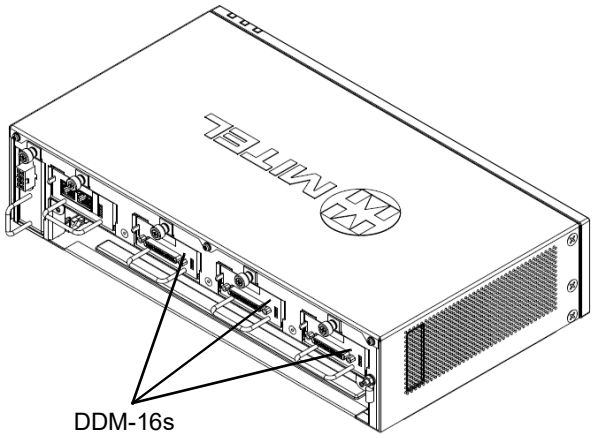
You can install up to three DDM-16s in the HX Controller chassis as described below. See **Figure 11** below for an illustration.

IMPORTANT	The DDM-16 can be removed with power on; however, you must wait for the REMOVE LED indicator to change to green before removing the module. Pressing and holding the release lever down until the LED indicator turns green allows the software to “disconnect” from the module. Failure to wait until the LED turns green, could result in pending information getting lost or unexpected system behavior. Other than the processor module, all other modules can reside in any bay. However, DB Programming will need to change, if you move a module (that was previously programmed) into a different bay location.

To install the DDM-16 in an HX Controller chassis:

1. Using a properly terminated, anti-static wrist strap, remove the module from its protective anti-static bag.
2. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops and the module release lever clicks into place (up position). Use just enough force to fully seat the connectors.
3. Tighten the fastener at the top of the module, securing it to the chassis. Do not over tighten this fastener.

Figure 11: 16-Port Digital Desktop Module (DDM-16) Installation



To remove a DDM-16:

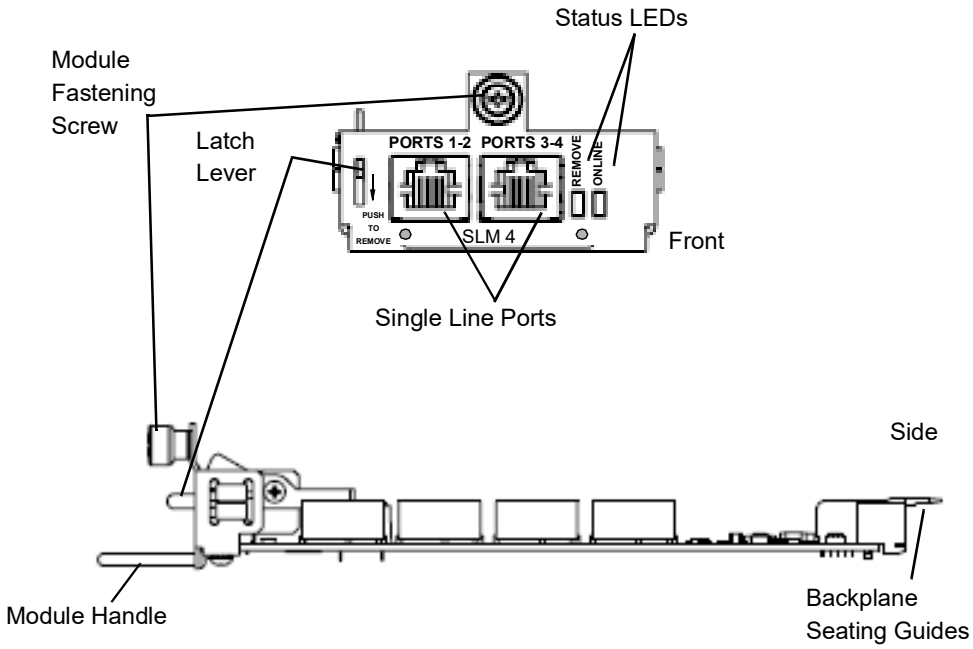
1. Using a properly terminated, anti-static wrist strap, loosen the fastener at the top of the module. This fastener secures the module to the chassis.
2. Push down and hold the module release lever.
3. When the REMOVE LED turns green, grasp the module handle and carefully remove the module from the chassis.

Installing a Four-Port Single Line Module (SLM-4)

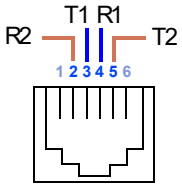
This section provides specifications, drawings, and procedures needed for installing an SLM-4, part number 580.2100.

Installing an SLM-4 consists of inserting the module into an empty bay on the back side of a CS Controller and assigning the module to the Uninstalled bay through DB Programming. You can then program up to four single line phones on the module ports. See **Figure 12**. Because the Auto Equip feature does not apply when installing an SLM-4, the module must be manually programmed in the system before it can be used.

Figure 12: Parts of the SLM-4, Front and Side Views



The pinout design for four-port and eight-port Single Line Module jacks (SLM-4 and SLM-8) is the same. Each RJ-14 jack represents two single line ports for connecting single line analog phones. The middle four pins in the RJ-14 connector are wired in accordance with Uniform Service Order Code (USOC) T568A specifications, as shown in the following diagram.



T1–R1 = Port 1 Tip and Ring
T2–R2 = Port 2 Tip and Ring

To install the SLM-4 module in a CS Controller:

1. Put on ESD (electrostatic discharge)-protective gear and ground yourself.
2. In DB Programming, uninstall any module that may be in the bay where you want to install the SLM-4.
3. Remove the physical module in accordance with the applicable procedure—trunk modules require the REMOVE LED to be lit before the module can be safely removed.

4. Insert the SLM-4 module into the empty bay. The green LED labeled ONLINE illuminates, indicating that the new hardware has been detected and the Single Line (SL) application is running.
5. Using the IP address of the Mitel 5000, open a DB Programming session to the system through the System Administration & Diagnostics application.

> **NOTE** The programming portion of the SLM-4 installation procedure may be performed in an offline session of DB Programming and then restored to the system using the Database Restore utility.

6. When DB Programming displays, in the left pane select System and then Communication Server.
7. Click the Uninstalled bay and click the down arrow.
8. Select **Single Line Module - 4**.
9. Program the SLM-4 module in the chosen uninstalled bay.
10. Click the SLM-4 folder and configure the ports.
11. Program the individual single line phones in DB Programming. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.
12. Connect and test the phones.

Installing the Processing Server (PS-1)

This section identifies the parts and processes, and describes the procedures for installing the optional Processing Server (PS-1). The PS-1 can be deployed using one of two methods:

1. as dedicated hardware
2. as a virtual appliance (vApp)

About the PS-1

As a dedicated hardware component, the PS-1 is a Linux server that connects through a local area network (LAN) to the HX Controller or CS Controller Base Server and provides call processing power for the expanded-capacity system. For a functional description, see “Processing Server Characteristics (Physical PS-1)” on **page 1-90**. For technical descriptions see the **Specifications** chapter and “Network Environment Requirements” on **page 1-307**.

VMware recently introduced the vApp, a software solution optimized for the cloud Operating System (OS). A vApp is a logical entity composed of one or more virtual machines, which can be managed as a unit. The vApp specifies and encapsulates components of an application (such as the PS-1) as well as the operational policies and service levels associated with it. The vApp gives application owners (such as Mitel) a standard way to describe operational policies for an application, which the cloud OS can consistently interpret and run. The vApp is a broad category of virtual applications that are built by independent software vendors (ISVs), system integrators, value-added resellers, and on-site IT administrators. See “Installing the Virtual Processing Server (vPS-1)“ on **page 205** for details about installing the PS-1 as a vApp.

By providing additional call processing power, the PS-1 (or vPS-1) allows the system to adequately support more phones than the Base Server chassis in stand-alone mode. vPS-1 performance will not to be any different to that of the Physical PS-1. The intention is to allow deployment onto a virtual machine, but not to achieve more BHCC, or extra resources, or to allow more phones to be connected.

Keyboards connected to a Physical PS-1 and keyboards used through a vSphere console will use U.S. keyboard mappings. If you are using a keyboard from a different region, be aware that some characters received at the PS-1 will not match what was pressed on the keyboard. The preference is to use an ssh connection to a PS-1 to enter or change values that depend on specific characters (for example changing a password or entering a command).

See the *MiVoice Office 250 Engineering Guidelines* for details on when a PS-1 (or vPS-1) might be needed.

Essential PS-1 BIOS Settings

The Basic Input/Output System (BIOS) settings of the PS-1 have been customized to control specifically the way the PS-1 software interacts with system hardware. Because the default BIOS settings have been modified to allow the system to operate, the BIOS settings must remain as they came from the factory.

NOTICE

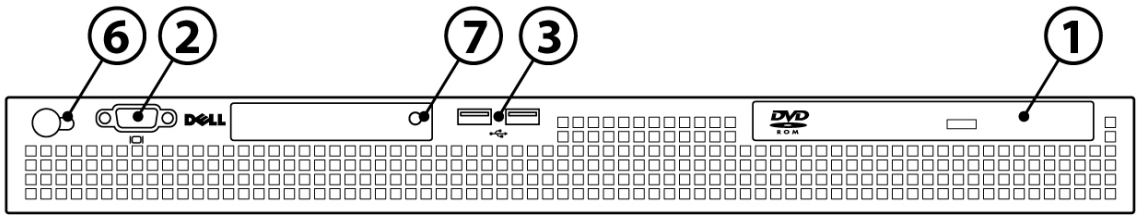
Incorrect or Defaulted BIOS Settings Will Make System Inoperable. If for any reason the BIOS settings of the PS-1 (or vPS-1) are defaulted or changed from the way they were shipped from the factory, contact Mitel Product Support to receive instructions for correctly reprogramming the BIOS.

PS-1 Hardware

This section provides front and rear view representations of the Dell PS-1 chassis.

NOTE The On/Off switch on the front of the chassis controls power to the system, and it should remain On at all times.

Figure 13: Processing Server (PS-1) Chassis, Front View



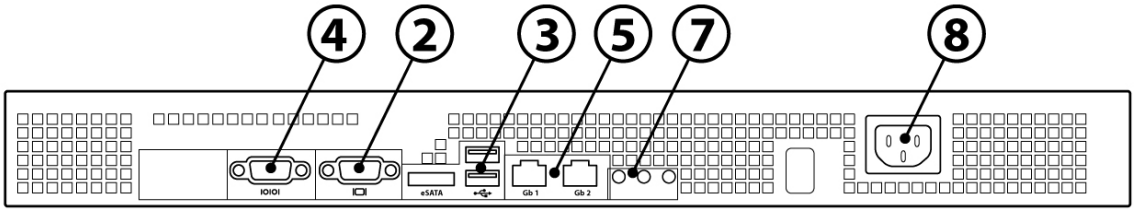
Item	Description
1	Digital Video Disk (DVD) drive. (Use for initial installation and system recovery.)
2	Video port. (DB-15 VGA connection to monitor. Can use front or rear ports.)
3	USB ports. The USB security key can be used in any USB port. (Use also to connect keyboard and mouse if necessary.)
6	Power LED.
7	System Identification LED. (Used for system location. See www.dell.com for details.)

NOTICE

Accidental System Reset. Do not mount the PS-1 in a rack directly above the Base Server. If you do, pulling down the front cover (not shown) of the PS-1 may cause the edge of the cover to press the three middle buttons on the LCD panel controls, causing the system to reset. If you mount the PS-1 in the same rack as the Base Server, make sure the hinged front cover of the does *not* open onto the front of the Base Server. Mitel recommends mounting the PS-1 below or away from the Base Server and Digital Expansion Interface (DEI) units.

Figure 14 shows the back of the PS-1 unit.

Figure 14: Processing Server (PS-1) Chassis, Back View



Item	Description
2	Video port. (DB-15 VGA connection to monitor. Can use front or rear ports.)
3	USB ports. The USB security key can be used in any USB port. (Use also to connect keyboard and mouse if necessary.)
4	Serial port. (Laptop connection using RS-232 cable/DB9M to DB9F straight-through connector, part number 813.1683.)
5	Ethernet ports.
7	System Identification LED. (Used for system location. See www.dell.com for details.)
8	Power cord plug.

Installing the Physical Processing Server (PS-1)

To see the location of connectors on the PS-1, see Figure 14 on **page 203** and Figure 15 on **page 211**.

To install a PS-1:

1. Rack mount the PS-1 (recommended).
2. Make sure the power switches on the front and back of the PS-1 are turned off. See Figures 14 and Figure 15 on **page 211**.
3. Connect the PS-1 to an AC power source, but do not turn either of the power switches on at this time. For electrical requirements, see Table 12, “PS-1 Specifications,” on **page 90**.
4. Using a CAT5 or CAT5e cable terminated on both ends with RJ-45 plugs, connect the PS-1 to the network through the same 100 Mbps or faster LAN switch as the Base Server.

   IMPORTANT The PS-1 and Base Server must be connected to the network on the same LAN, VLAN, and subnet. For additional details, see “Network Environment Requirements” on **page 1-307**.

5. Connect a monitor and keyboard to the PS-1. If a monitor and keyboard are not available, you can use a laptop PC equipped with a DB-9 serial port. In the alternate mode using a laptop PC, connect the PS-1 to the laptop with a straight-through RS-232 cable and null modem with hardware flow control, as shown in **Figure 15**.
6. Turn power on using the On/Off switch on the back of the chassis. Leave the On/Off switch on the back of the chassis turned on at all times.
7. Turn power on using the button switch on the front of the chassis. Use the button switch to control power to the system. Leave the On/Off switch on the back of the chassis turned on at all times.

To associate the Base Server and the PS-1 on the LAN:

Refer to page 325 under the “Expanding by Adding a PS-1” section for the procedure to associate the Base Server and the PS1 on the customer’s network.

To configure the PS-1 on the LAN:

Refer to page 326 under the “Expanding by Adding a PS-1” section for the procedure to configure the PS-1 on the customer’s network.

Installing the Virtual Processing Server (vPS-1)

The Open Virtualization Format (OVF) standard 1.0 is employed to represent a single virtual machine or combination of multiple virtual machines. Note that OVF files are recognized by their *.ova* file extension. The vPS-1 is a virtual appliance (vApp) which is deployed through vCenter Server with the vSphere Client. The PS-1 vApp runs on vSphere 5.1, or 5.5.

Note: This guide does not describe the setup and operation of the VCenter Server with the VSphere Client. Refer to the VMware documentation for details.

For a list of servers that are approved for use with product releases 5.1 and 5.5, consult the VMware Compatibility Guide (http://www.vmware.com/resources/compatibility/).

Installing and activating vPS-1 software consists of the following steps:

1. Logging in directly to the ESXi hosts or to the vCenter Server using the vSphere Client application.
2. Deploying vPS-1 as a virtual machine and virtual application operating on VMware vSphere 5.1, or 5.5.
3. Powering up the Virtual Machine and commissioning vPS-1.
4. Configuring and adding a new USB port and license key.

Before you Begin

The following conditions must be in place before you begin the installation process:

- VMware ESXi release 5.1, or 5.5 installed on an approved server.
- An approved server as follows (consult the VMware Compatibility Guide (http://www.vmware.com/resources/compatibility/)):
  - Minimum 1 GB RAM
  - 80 GB free disk space.
  - Single socket/single core processor
  - Network adapter
  - One USB port
- vSphere Client installed on a Windows workstation. Refer to the VMware website for detailed installation procedures and additional documentation.
- (Optional) vCenter Server deployed on the network. Refer to the VMware website for detailed installation procedures and additional documentation.
- A DNS server that is reachable from the platform.
- vPS-1 software is contained in an Open Virtualization Alternative (OVA) format archived file. This archive OVA file contains the OVF 1.0 descriptor and VMDK file.
- Verify that the correct version of vPS-1 software is being deployed, based on the Base Server version, and compatible applications.

Go to Mitel Online at https://www.ebiz.mitel.com. Point to **Support**, **Technical Support**, then click **Software Downloads**.

Install the PS-1 Virtual Software

NOTE This installation process assumes that a virtual machine has been created using the Debian GNU/Linux version 4 - 32 bit OS.

1. Retrieve the PS-1 VMware OVA archived file from Mitel on-line. Make note of where the file has been saved.
2. Launch the vSphere Client.
3. Click **File ->Deploy OVF Template. . .**.The *Deploy OVF Template* screen displays.
4. In the **Deploy from a file or URL** drop down list, do one of the following:
   - If the OVF template file was downloaded to the local computer or to a network share drive, then click **Browse** to locate the file.

     NOTE On Microsoft Vista operating systems, select *.ova* in the **File Type** drop down menu.

     CAUTION

     Do not install vPS-1 from any CD ROM drive that may be attached to the virtual machine. The CD ROM interface is not present in the vSphere Client and should not be added since it may impact the overall performance of the system.
   - If the OVF template file is on the internet or accessible through a web browser; enter the URL of the location of the file.
5. Click **Next**. The *OVF Template Details* screen displays.
6. Click **Next**. The end user license agreement screen displays.
7. Click **Accept** to accept the license agreement, then click **Next**. The *Deploy OVF Template Name and Location* screen displays.
8. Enter a meaningful name for vPS-1, or accept the default name.
   Click **Next**.

   NOTE When deploying vPS-1 in a vCenter environment, the wizard may additionally prompt for a Rsource Pols, Datastore, and Network Mapping if several options are available. Please contact your Data Center administrator for more details on which Datastore or Network Mapping to use.
9. If the cluster to which the OVA file is being deployed contains a resource pool, then the *Deploy OVF Template Resource Pool* screen appears. Otherwise the *Deploy OVF Template Ready to Complete* screen appears.
   Select the appropriate resource pool and click **Next**. The *Deploy OVF Template Storage* screen appears.
10. Select the Datastore where the virtual machine files will be stored. This option only appears if multiple Datastores are available on the destination cluster or host

Select a datastore large enough to accommodate the virtual machine and all of its virtual disk files.
Click **Next**. The *Deploy OVF Template Disk Format* screen appears.

11. Click **Thick Provision Lazy Zeroed**.
    Click **Next**.
12. If the network defined in the OVF template doesn't match the name of the template on the host to which you are deploying vPS-1, you are prompted to configure Network Mapping. Contact your Data Center administrator for more details about which Network Mapping to use.
13. Click **Next**. The *Deploy OVF Template Deployment Ready to Complete* screen appears.
14. Review the information.
    Select **Power on After Deployment**.
    Click **Finish**.
    vPS-1 will be generated and deployed on the server, and the virtual machine will be powered up.

Time: This process may take up to 10 minutes to complete depending on network traffic and the performance of the server.

15. When the dialog indicating that the deployment is complete appears, click **Close**.
    vPS-1 appears in the inventory list in the left side navigation pane.
16. Click on the newly created vPS-1.
17. When the vPS-1 has powered up, you can begin configuring the USB port to accommodate the HASP license key.

IMPORTANT	The PS-1 and Base Server must be connected to the network on the same LAN, VLAN, and subnet. For additional details, see “Network Environment Requirements” on **page 1-307**.

To associate the Base Server and the PS-1 on the LAN:

Refer to page 325 under the “Expanding by Adding a PS-1” section for the procedure to associate the Base Server and the PS1 on the customer’s network.

To configure the PS-1 on the LAN:

Refer to page 326 under the “Expanding by Adding a PS-1” section for the procedure to configure the PS-1 on the customer’s network.

Configuring and Adding a New USB Port and License Key

1. Insert the HASP license key into a spare port on the VMware ESX server.
2. Select the vPS-1 virtual machine from the inventory list in the left side navigation pane.
3. Right-click and choose **Edit Settings**, or select **Edit virtual machine settings** from the **Basic Tasks** list on the right navigation pane.
4. Select the **Hardware** tab and click **Add. . .**.
   The *Device Type* screen displays.
5. Select **USB Device** and Click **Next**.
   The list of connected USB devices will appear.

6. Select the HASP license key and Click **Next**.
7. Confirm the information is correct and Click **Finish**.

The new license can be displayed in the Administrator Web Portal, in System Administration and Diagnostics, and in Database Programming.

In the Administrator Web Portal, there is a new parameter on the License Details page: Virtualized PS-1 Support.

In System Administration and Diagnostics, there is a new parameter on the License Information window: Virtualized PS-1 Support.

In Database Programming, the new parameter, Virtualized PS-1 Support, is displayed in the About window. A full list of software licenses is also displayed in Database Programming under **Software Licenses**.

IF there is no vPS-1 license on a system installed with a vPS-1, the system will reboot every 4 hours.

If there is no HASP key on a system installed with a vPS-, the system will reboot every 4 hours.

DIGITAL EXPANSION INTERFACE (DEI)

The following sections provide processes and procedures required for installing digital interface equipment.

DEI Planning and Cabling

This section describes installing the Mitel Digital Expansion Interface (DEI) unit, the Digital Endpoint Module (DEM-16), and the eight-port Single Line Module (SLM-8). Other digital telephone-related items are also discussed such as Single-Line Adapters (SLA) and supported digital telephone installations. Information on installing an external voice processing system is also discussed.

Installation Summary

Follow the new installation summary guidelines with any additions mentioned here. See “Pre-Installation Checklist” on **page 1-133**.

- Run sufficient lengths of cable to any newly installed digital telephones and to optional equipment for connection to the MDF. See page 152. Allow for adequate strain relief.
- Terminate the station cables on modular jack assemblies [line jack units] at the phone locations. See **Table 35** for loop limits.

Table 35: Loop Limits for Digital and Single Line Phones

Phones	Port Impedance	Loop Limit (24 AWG)
From Base Server chassis: • To a single line phone or playback device connected to the built-in port • To a single line phone or playback device driven by a SLM-4 port From DEI chassis: To a single line phone or playback device driven by a SLM-8	600 Ω	2000 ft (609 m) REN 3.0
From **DDM-16 (HX only)** or DEI chassis: • To digital telephones • To a digital telephone equipped with PCDPM and MDPM and connected to PKM/DSSs or SLA	62 Ω	1200 ft (366 m)
From an MDPM: To a single line phone	195 Ω	3800 ft (1158 m)
From an SLA: To single line phones or playback devices. Supports facility code OL13C.	800 Ω	15,560 ft (4742 m)

- On the MDF backboard, assemble any additional terminal blocks for digital telephones, including SLAs. See page 153.
- Install the DEI chassis near the MDF in a Mitel or other rack. Also, mount the chassis in the rack at a height appropriate for reading the Status panel. Allow room above and below the chassis for later expansion. If wall-mounting is desired, contact the Mitel DataNet CommSource Division for information.

  See Figure 6 on **page 221**, Figure 6 on **page 221**, Figure 7 on **page 222**, Figure 8 on **page 225**, and Figure 9 on **page 226** for examples.

- Install the DEM-16(s) in the chassis—see page 228.
- Power up the system—see page 247.

Running Cables to Phones

Carry out the following guidelines when running cable for digital telephones:

- Develop floor plans to aid in proper digital telephone cabling in a star (home run) configuration from the DEI chassis. If necessary, run the necessary cabling to where the digital telephone will be used. See “Cabling to Phones” on **page 1-65**.
- If desired, two- or three-pair twisted cable and four- or six-conductor modular jacks [line jack units] may be used for *all* digital and analog phone connections. This allows the various phone types to be easily interchanged, if necessary. (If three-pair cable is used with a U.S. system, extra terminal blocks and the use of cross-connect wiring techniques are required.)
- Install proper type cable for the application according to the National Electrical Code and local building codes [BS 6701 in Europe].
- Avoid cable runs parallel to fluorescent light fixtures or AC lines not in conduit. If these obstacles are unavoidable, run the cables across them at right angles.
- Do not run cables inside electrical conduit already occupied by AC power cable. (To do so is a violation of the National Electrical Code or BS 6701.)
- Do not run cables near equipment with electric motors or through strong magnetic fields, such as those generated by large copy machines, arc welding equipment, heavy motors, etc.
- Do not place station cables where they can be stepped on or where they can be rolled over by office furniture.
- If using multi-pair (for example, 25-pair) cable runs to multiple station locations, do not include AC-ringing single line phones, or AC-ringing auxiliary equipment.
- Using 24 AWG solid copper cable, do not exceed the loop limit measurements for the station cable lengths. The resistance values (in ohms) are loop measurements; the distance values (in feet or meters) are the maximum one-way measurements from the equipment chassis. The loop limit for digital telephones is 62Ω and 1200 ft (366 m). See Table 35 on **page 209** for additional loop limits for devices connected to the MiVoice Office 250.

Because the MiVoice Office 250 uses digital voice and data transmission between the digital telephones and the chassis, the following cabling requirements apply:

- Do ***not*** use shielded cable, and do ***not*** use cable smaller than 24 AWG.
- Do ***not*** parallel cables (for example, do not “double up on the copper”).
- Do ***not*** “bridge” or “Y” off the cable to another location.
- Digital telephones do ***not*** need to be routed in separate multi-pair cable runs. For example, if using 25-pair cable runs to multiple station locations, it is acceptable to include single line phones, auxiliary equipment, and trunks in a cable being used for phones.

Terminating Cables at Digital Telephone Locations

Terminate the digital telephone, single line phone, and playback device station cables on four-conductor modular jack [six-conductor line jack unit in Europe] assemblies at the station locations.

Do not mount the assemblies on the wall at this time; they will be wall-mounted later when the station instruments are installed. See **Figure 15** and Figure 16 on **page 212**.

Newer 8528 and 8568 Digital Telephones

Unlike the 8520 and 8560 digital telephones, the newer 8528 and 8568 telephones use standard RJ11 wiring configuration as defined in Telecommunications Industry Association (TIA) 968. See Table 22 on **page 112** for cable termination information.

Other Digital Telephones

IMPORTANT If tip and ring are reversed when installing a telephone, the telephone will ***not*** function.

Figure 15: Modular Jack Assembly Wiring Diagrams (U.S. System)

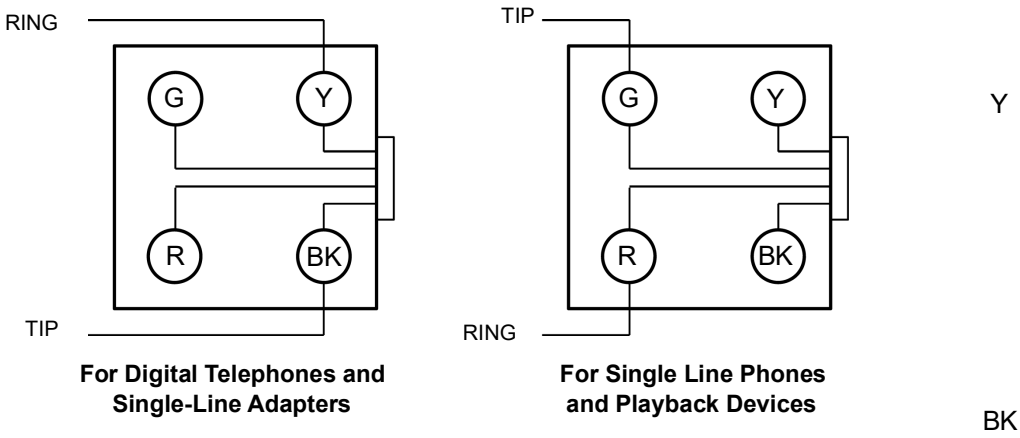
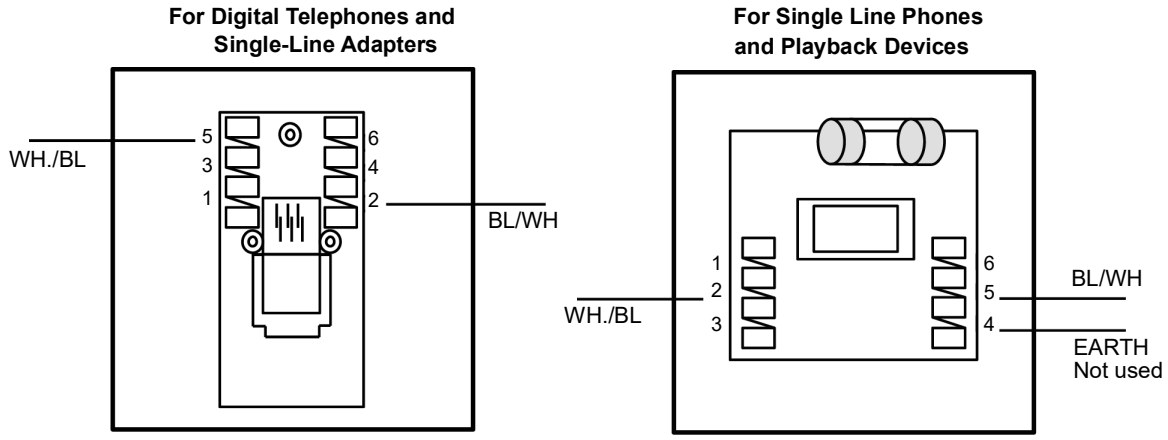


Figure 16: Line Jack Unit Assembly Wiring [European System]



Labeling Station Cable and Mapping Circuit Numbers

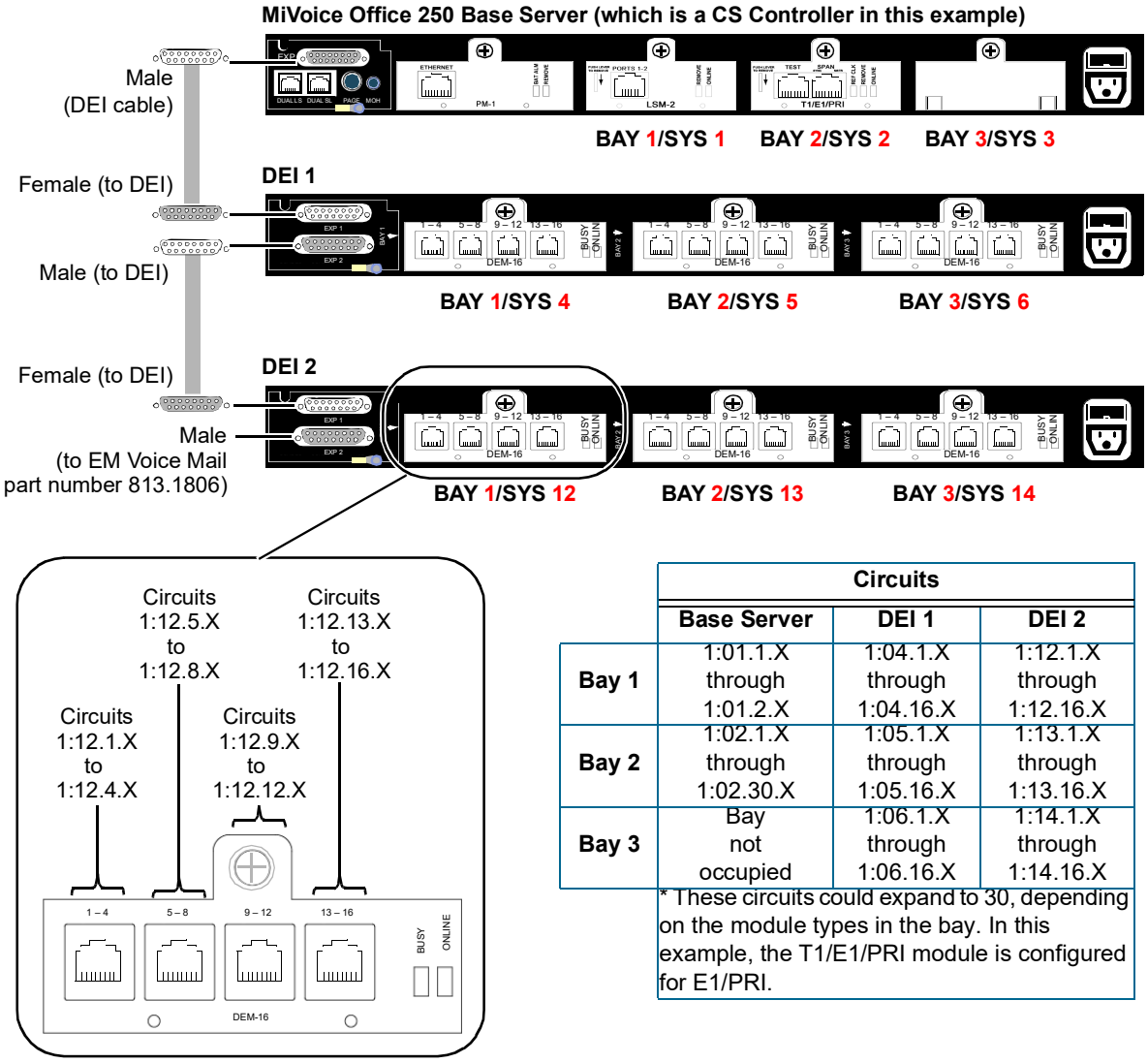
Label both ends of each cable with the station circuit number (hardware address) and how it relates to the corresponding location in the DEI chassis. All circuit numbers are preceded with a two-digit node number. For example, 02.02.16.01 is a port on node 2, module 2, circuit 16.

In some circumstances, a station circuit can have more than one purpose. For example, two single line devices can be installed on a single DEM-16 circuit. The first device is assigned circuit number XX.YY.01, and the second device is assigned circuit number XX.YY.02. Similarly, a PC Data Port Module and Modem Data Port Module can be assigned circuit numbers. In this case, the phone is assigned circuit number XX.YY.01, the first data port is assigned number XX.YY.02, and the second data port is assigned number XX.YY.03. For detailed information about programming phones, refer to the *MiVoice Office 250 Features and Programming Guide*.

The default system assigns extension number 1000 to station circuit 01.01.01, extension number 1001 to station circuit 01.02.01, and so on to match the initial configuration. Circuit 01.01.01 is equipped as an administrator and as the primary attendant phone when the system is in the default configuration. Through DB Programming, the circuits can be assigned 1–5 digit extension numbers.

Figure 17 on **page 213** illustrates how the Bay numbers in the CS Controller and each DEI correspond to the System Board/Circuit information in Message Print. This CS Controller example only shows two DEIs, but you can have up to four DEIs in a system (see Figure 7 on **page 222**). Table 36, “System Board:Bay:Application Relationship in Message Print (CS Controller Example),” on **page 214** summarizes the system board/bay/application relationships between a MiVoice Office 250 and Message Print.

Figure 17: DEM-16 Circuit Mapping Example



	Circuits		
	Base Server	**DEI 1**	**DEI 2**
Bay 1	1:01.1.X through 1:01.2.X	1:04.1.X through 1:04.16.X	1:12.1.X through 1:12.16.X
Bay 2	1:02.1.X through 1:02.30.X	1:05.1.X through 1:05.16.X	1:13.1.X through 1:13.16.X
Bay 3	Bay not occupied	1:06.1.X through 1:06.16.X	1:14.1.X through 1:14.16.X

* These circuits could expand to 30, depending on the module types in the bay. In this example, the T1/E1/PRI module is configured for E1/PRI.

LEGEND:

BAY 1/SYS 4 = Bay # in the chassis is the same as equivalent System # in Message Print

= DEI Cable (part number 813.1847 or 813.1847-001)

Table 36: System Board:Bay:Application Relationship in Message Print (CS Controller Example)

		Relates to this bay/application in . . .	
		5000 CP v1.x System	5000 CP v2.x System
System Board (Module) in Message Print . . .	1	Bay 1: LSM-2, T1M or T1M-2	Bay 1: LSM-4, T1M or T1M-2, BRM-2
	2	Bay 2: LSM-2, T1M or T1M-2	Bay 2: LSM-4, T1M or T1M-2, BRM-2
	3	Bay 3: LSM-2, T1M or T1M-2	Bay 3: LSM-4, T1M or T1M-2, BRM-2
	4	On-Board LS-2 Application	DEI-1, BAY1
	5	On-Board SL-2 Application	DEI-1, BAY2
	6	Not used	DEI-1, BAY3
	7	IPRA7 Application (All IP phone/networking call control)	IPRA7 Application (All IP phone/networking call control)
	8	Rch (Resource Application)	Rch (Resource Application)
	9	IPRA9 Application	IPRA9 Application
	10	Not used	On-Board LS-2 Application
	11	Not used	On-Board SL-2 Application
	12	Not used	DEI-2, BAY1
	13	Not used	DEI-2, BAY2
	14	Not used	DEI-2, BAY3
	15	Not used	Not used
	16	Not used	Not used

Unpacking the DEI Equipment

After all digital telephones and related devices are installed and station/device cabling has been run or is in place, you are now ready to install the DEI chassis and modules.

NOTICE

Electrostatic Discharge (ESD). System modules and components are static-sensitive. Handle the modules by the edges only and keep them in their protective bags until they are to be installed. Do not bend or touch the component pins or subject them to a static charge. When working with the modules, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the module during shipping. Any static charge—no matter how small—must be discharged from the body before touching the modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will not be replaced.*

To unpack the equipment and inspect the equipment chassis and all accompanying parts:

1. Check the parts against the packing slip and inspect them for damage. If any item is missing or damaged, contact Technical Support immediately. See Table 57, “Contact Information,” on **page 370**.
2. Check all circuit module assemblies as follows:

a. Check each module to ensure there is no shipping foam or tape attached.

b. Inspect each module for shorted components.

c. Return all modules to their protective bags until they are ready to be installed in the chassis.

3. Check that the protective covers over the module bays are in place. These covers protect debris or other foreign matter from entering the system and possibly harming the internal components. These covers should remain in place until you are ready to install a module.

Installing DEI Hardware

IMPORTANT Do *not* plug in any power cords until instructed to do so. Powering up the system occurs later in the installation.

Total digital telephone capacity is 192 (12 DEMs in four DEI chassis). See “Capacities” on page 67 for additional system capacity information.

The following items are discussed in this section:

- Considerations on **page 216**
- Installing/Upgrading the DEI Chassis on **page 217**
- Troubleshooting on **page 218**
- Downgrading DEI Chassis on **page 219**
- Time Slots for DEIs on **page 219**

Considerations

In addition to any hardware/software information discussed in this manual and in the DEI notice (part number 835.3242) that ships with the DEI unit, please take into consideration the following, when installing and bringing up DEI units:

- If adding DEIs to an existing system, perform a database backup and database save before installing any DEIs. Be prepared to log on and view the logs.
- Use the Power Calculator (available on the system software DVD) to approximate the power usage for the system chassis and related peripherals, including the DEI chassis. For the purposes of the calculator, the DEI chassis is classified as a Small App.
- In addition to the installation guidelines for hardware earlier in this manual, check for proper grounding. For example, the ground lug on the chassis must be in parallel with the AC ground via the power supply. Check with local Building Safety Inspectors on the installation of the ground wire for the system. Conduit may be required between the ground bar and earth ground. Refer to the System Equipment Grounding Guidelines, part number 835.2845, for additional information.
- Detach the voice mail system from the last DEI in the chain before adding additional DEI units. When installing and upgrading new DEIs, leave the system powered on, disconnect existing DEIs, and connect new DEIs one at a time. If the system has DEIs, disconnect them, and connect only DEI #1. If no DEIs exist, and you are adding new DEIs, connect only DEI #1.
- If the attached DEIs do not come online, recycle the power on all DEIs; also, check that you selected the correct mode (2- or 4-mode) in DB Programming. if they remain offline, contact Technical Support for further assistance.

  The DEI low-level firmware controls how it communicates. For this upgrade, the method of communication changes. To update the DEI low-level firmware, it must be communicating in the previous two-DEI configuration. The MiVoice Office 250 software automatically loads the latest DEI software when DEIs are connected and the system is restarted.

- When adding new DEI units, you can connect two DEI cables together to extend the reach between the MiVoice Office 250 and the DEI units you want to add. This allows existing DEIs to remain in place while you upgrade the new DEIs.
- When installing any DEI unit, make sure module bays are empty. DO NOT attach DEIs to the system with populated module bays. The servers are shipped without modules installed. The examples throughout this document are for illustrative purposes only. Module covers must remain in place if there is not an actual module to occupy a module bay. The covers help to ensure foreign material does not enter the chassis.
- In 4-DEI mode, there are 48 voice channels to each DEI; this provides one voice channel per digital telephone. Features that require a second voice channel, such as OHVA, MDPM, and dual-SLA, require special attention. Enable these features using the current methods; however, these features use a voice channel that could otherwise be used by a telephone on the same DEI. DB Programming counts the time slots used by these features and enforces the maximum timeslot limit for each DEI.

Installing/Upgrading the DEI Chassis

To install the DEI chassis:

1. Make sure you have upgraded system software to v3.0 or later.
   - *If the system has DEIs*, disconnect them, and connect only DEI #1. If a voice mail system was connected to the last DEI, remove the VM—DEI cable from the last DEI in the chain.
   - *If no DEIs exist*, and you are adding new DEIs, connect only DEI #1.

   Refer also to the DEI notice, part number 835.3242, that is included in the box with the DEI you want to install.
2. Install the chassis in the rack under the MiVoice Office 250 Controller. Allow room below the chassis for future expansion (for example, more DEI chassis or an optional external voice processing system) and proper cooling. See Figure 5 on **page 220**, Figure 6 on **page 221**, Figure 7 on **page 222**, and Figure 8 on **page 225** for examples.

   NOTE Prior to the 5000 Release v6.0, the optional external voice processing system would have been the Voice Processing Unit (VPU — unsupported since v3.0) or Enterprise Messaging (unsupported as of v6.0). Beginning with 5000 Release v6.0, NuPoint Unified Messaging (UM) is the supported external voice processing system.
3. *When installing one or more DEIs, do the following:*
   a. Leave the system powered on. **Do not power cycle the Base Server or PS-1**, if applicable.
   b. Disconnect any existing DEI units.
   c. Install the first DEI unit and connect the DEI cable to the system. After the first DEI is installed and upgraded, power cycle the DEI, and then remove the cable and install the next DEI. You can connect several DEI cables together as shown in Figure 7 on **page 222** to extend the cable and reach other DEI units.
   d. After the last DEI is upgraded, reconnect all DEIs and power up the system. All are now functional and ready for programming. Keep in mind when programming and using more than 2 DEI, you must select the correct mode—2-mode or 4-mode.

To verify functionality and a successful upgrade:

1. Insert one DEM into each chassis and equip the DEM in DB Programming.
2. Check the log file in the Admin Web Session pages.
   a. Establish an Admin Web session. Log on using the appropriate user name and password.
   b. Click the **Logging** tab, then click the **Log Files** link in the navigation pane.
   c. Locate the DEI Logs section and click **`romd_out_dei.log`**.
   d. Locate the following line. This line indicates a successful DEI update.

      `"update_dei: Update done size = 65536, Reloading Prom"`

      An alternative method to check for when a DEI upgrade is complete is to look in Message Print for the Digital Keyset Board online message. This method assumes that at least one DEM module is inserted in the DEI and it is equipped in DB Programming.
3. Do either of the following:
   - Begin programming the system. Refer to the *MiVoice Office 250 Features and Programming Guide*, part number 580.8006.
   - Continue to installing other optional equipment:
     - NuPoint Messenger: Refer to the *MiVoice Office 250 and NuPoint UM Integration Guide*, part number 580.8008
     - External USB modem (page 328)

Troubleshooting

See **Table 37** for information on troubleshooting DEIs.

Table 37: Troubleshooting DEIs

Scenario	Cause	Resolution
DEI is going offline	Check the Admin Session Web page log: `romd_out_dei.log`. If you see multiple messages, such as “MP present,” this is an indication the DEI link is not stable.	• Make sure cables are firmly attached. • Try a different DEI cable. • Connect one DEI only. Do this for all DEIs. This will help to isolate the faulty device. • Make sure the system is grounded properly.
After upgrading to v3.0 or later, the system cannot communicate with one or more DEI(s).	The DEI low-level firmware was not upgraded correctly.	Upgrade one DEI at a time as described in the previous pages. Check the Admin Session Web page log `romd_out_dei.log` for successful upgrade information.

(Sheet 1 of 2)

Table 37: Troubleshooting DEIs (Continued)

Scenario	Cause	Resolution
After upgrading to v3.0 or later, the system cannot communicate with the third and/or fourth DEI(s).	Incorrect license.	Obtain a valid v3.0 license with the appropriate number of DEIs.
	One or more DEI(s) have over allocated voice channels.	Redistribute devices among the DEIs to avoid voice channel over allocation. Disable features such as OHVA, dual-SLAs, or MDPMs to reduce the required voice channels on a DEI.
	The system is set to 2-DEI mode.	Change the DEI Mode setting in DB Programming to 4-DEI mode.
The system is showing “Out of Voice Channels” alarm.	A call on a DEI failed because of timeslot/voice channel over allocation.	Move devices to other DEIs to eliminate the timeslot over allocation.
(Sheet 2 of 2)		

Downgrading DEI Chassis

To downgrade DEI chassis:

To connect a DEI that has v3.0 or later firmware to a pre-v3.0 5000 CP, the system must downgrade the DEI firmware during installation. Do not connect more than one DEI to the pre-v3.0 5000 CP at a time. Follow the procedure “Installing/Upgrading the DEI Chassis” on **page 217**.

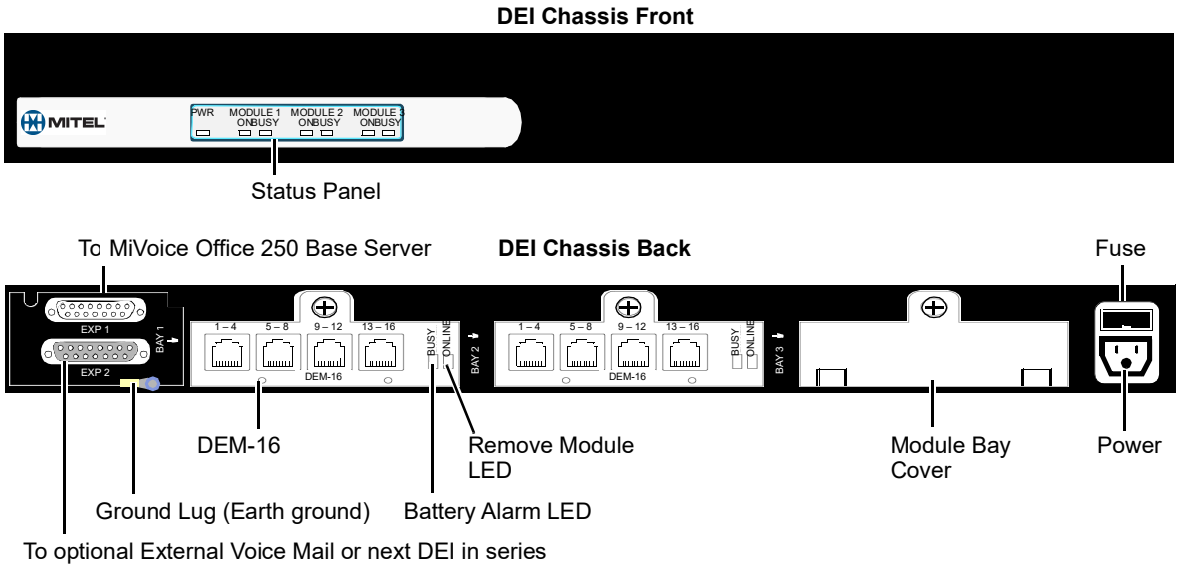
Time Slots for DEIs

Table 38 summarizes the devices, features, and modules that require time slots.

Table 38: Time Slots for DEIs

Device/Feature	Time Slots Used
Digital Telephones	1
Digital Telephone/MDPM	2
Dual Single Lines	2
Dual DID	2
Single Line/DID	2
Digital Telephone with OHVA Receive On	2
Single Line Module - 8	8

Figure 5: Example MiVoice Office 250 DEI Chassis Front and Back View



NOTE Prior to the 5000 Release v6.0, the optional external voice processing system would have been the Voice Processing Unit (VPU — unsupported since v3.0) or Enterprise Messaging (unsupported as of v6.0). Beginning with 5000 Release v6.0, NuPoint Unified Messaging (UM) is the supported external voice processing system.

Figure 6: Example DEI Connection to MiVoice Office 250 Base Server: 2-DEI

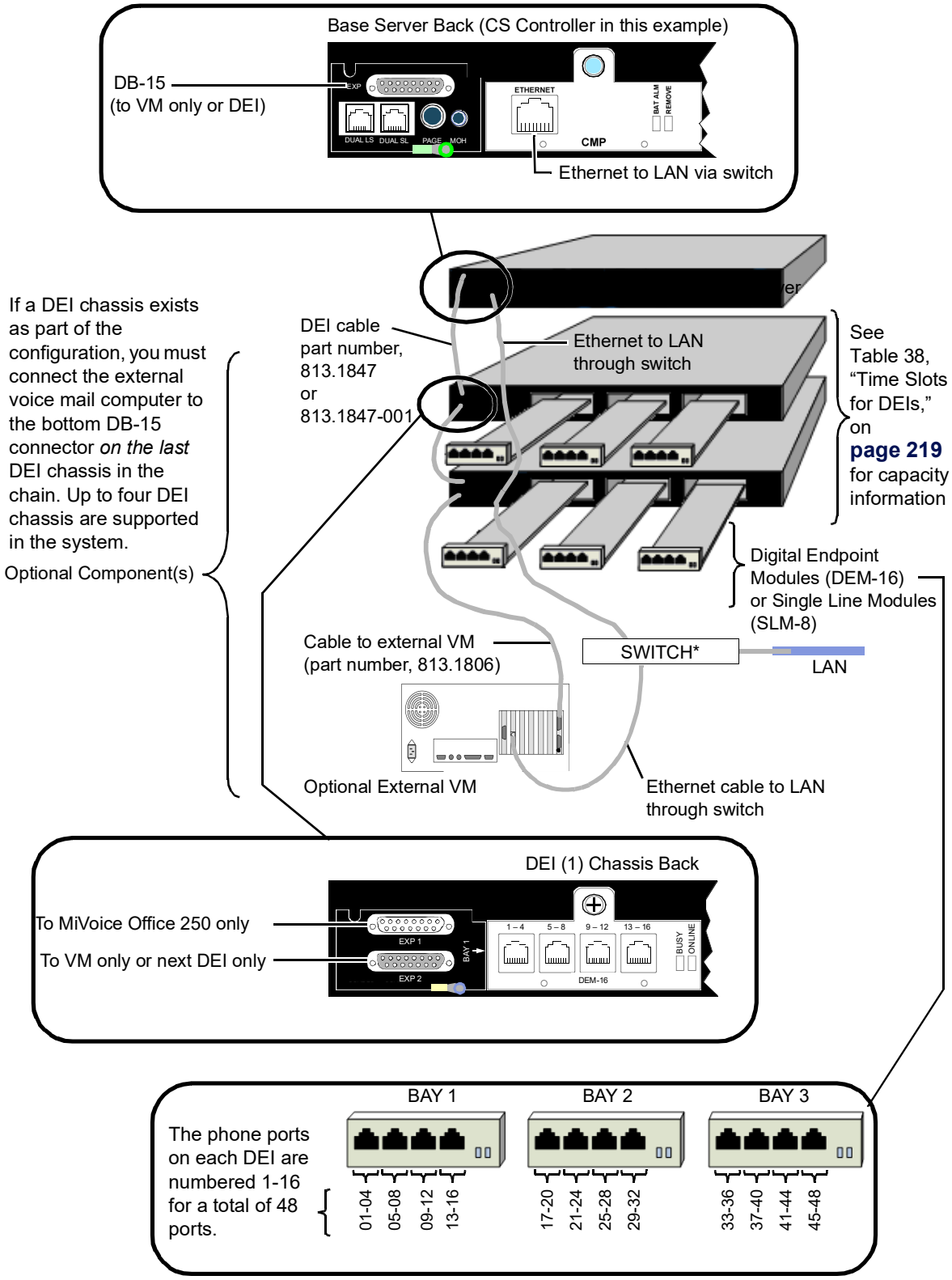
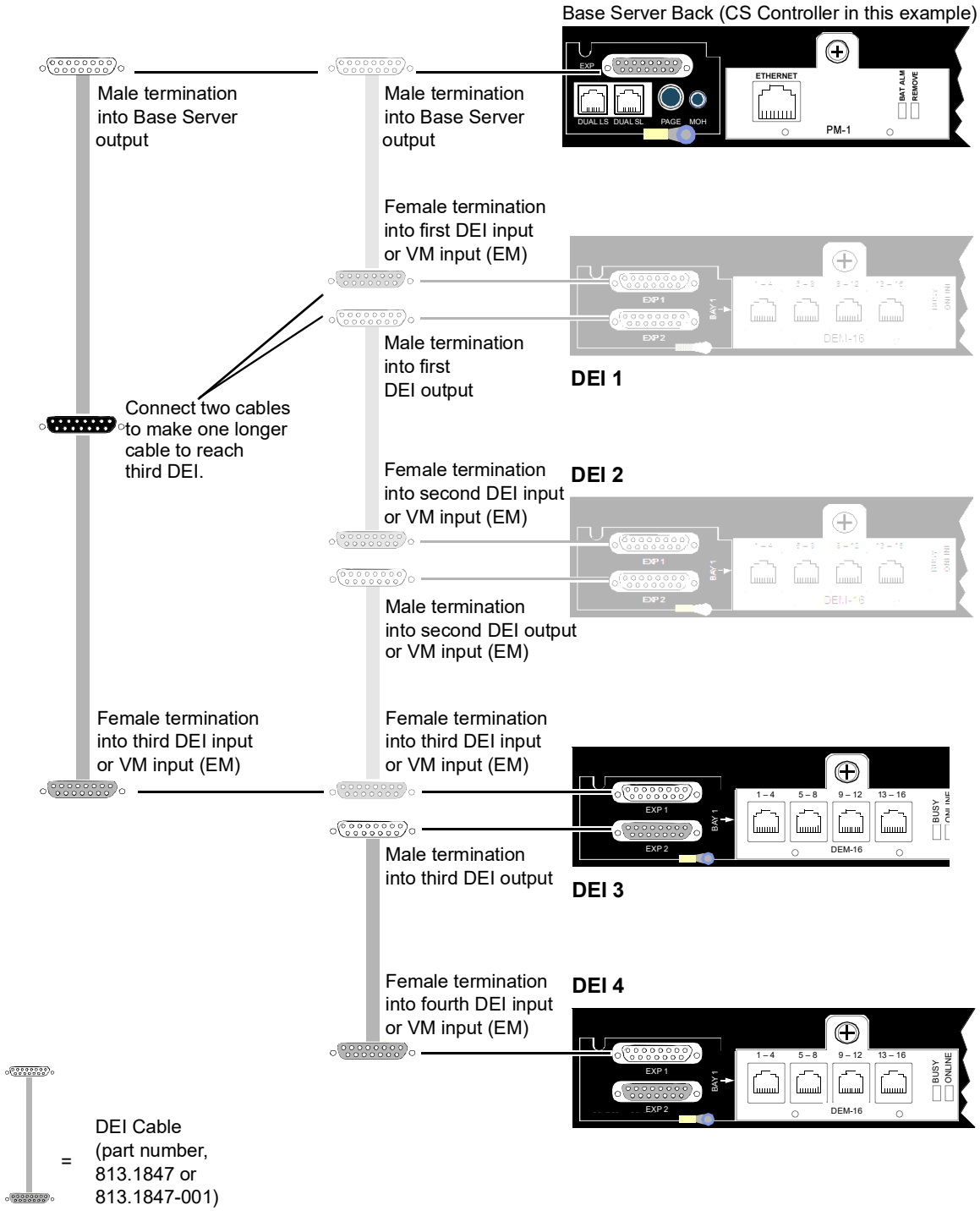


Figure 7: Example Adding DEIs to the MiVoice Office 250 Base Server: 4-DEI

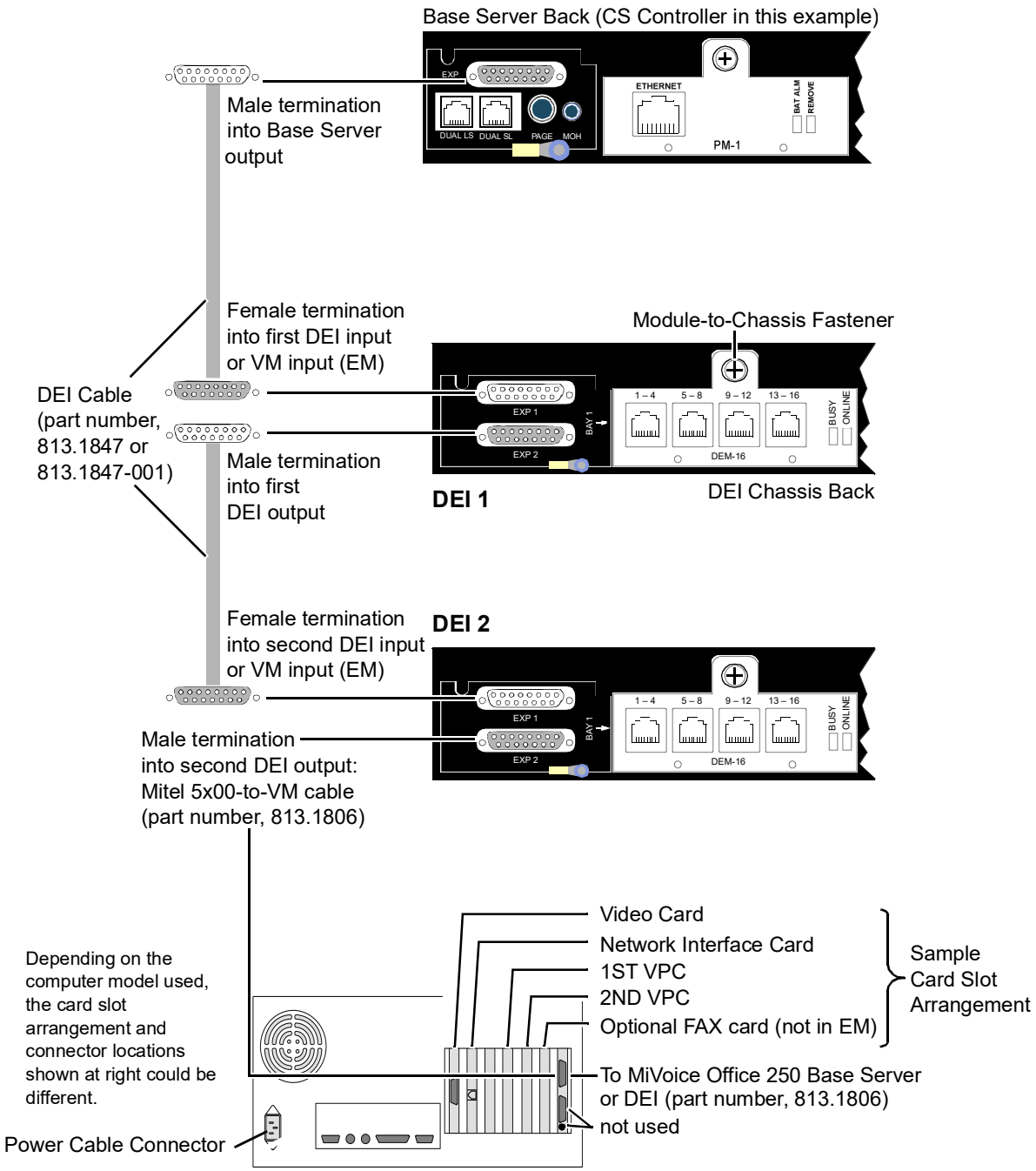


Attaching DEI Units and External Voice Mail System

See Figure 8 on **page 225** and Figure 9 on **page 226** for information on attaching the DEI to an external voice mail system.

NOTE Prior to the 5000 Release v6.0, the optional external voice processing system would have been the Voice Processing Unit (VPU — unsupported since v3.0) or Enterprise Messaging (unsupported as of v6.0). Beginning with 5000 Release v6.0, NuPoint Unified Messaging (UM) is the supported external voice processing system.

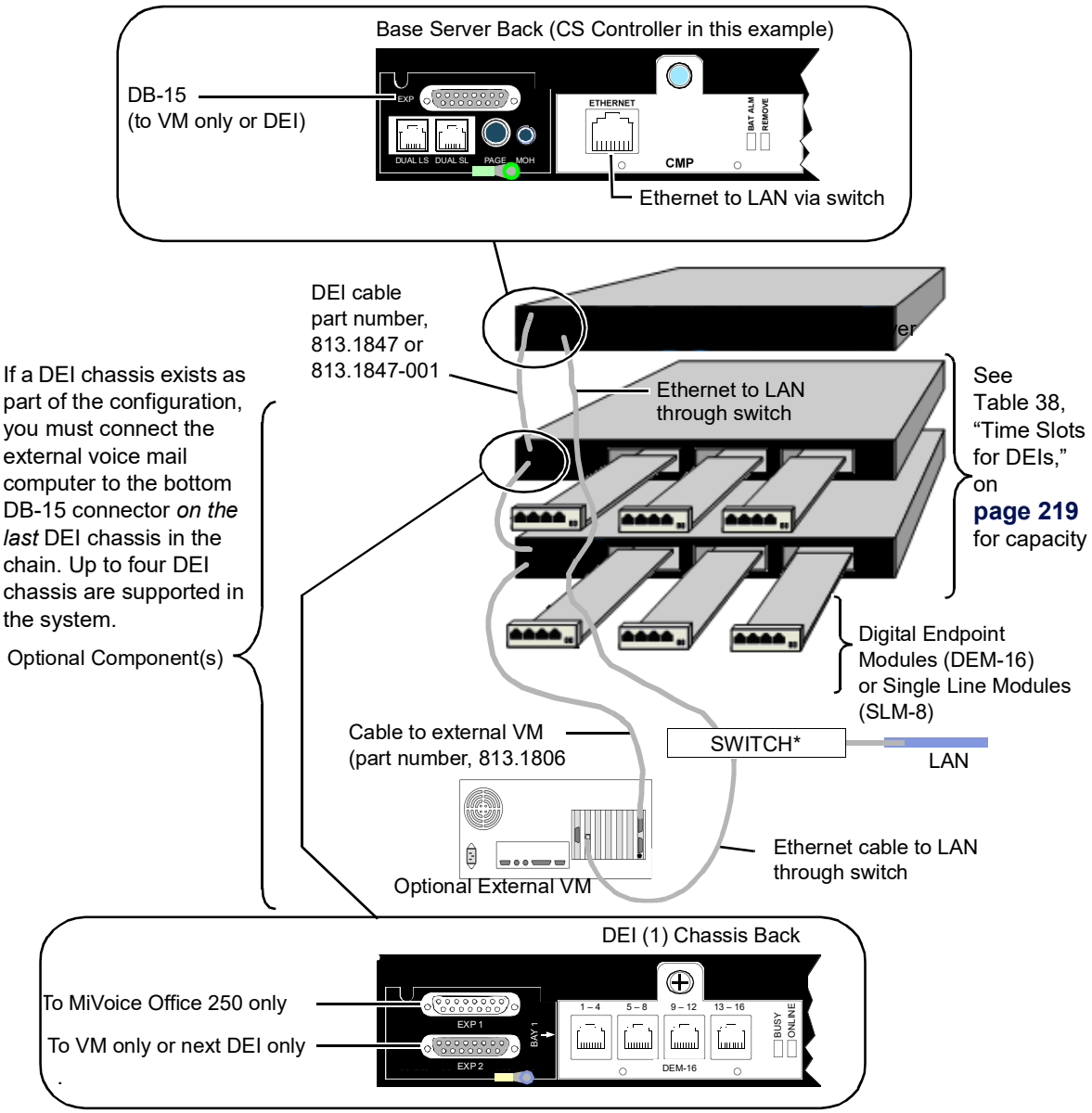
Figure 8: Example DEI and External Voice Mail Connection to Base Server



NOTICE

Equipment Damage: Use adapters when connecting any 6-pin plug to an 8-pin jack. Although a 6-pin plug will fit into an 8-pin jack, the plastic on the 6-pin plug housing will damage pins 1 and 8 on the 8-pin jack. Not using adapters will increase the likelihood of failure (to pins 1 and 8 in the 8-pin jack) should that jack need to be used with an 8-pin plug in the future. Various adapter types are available based on how you choose to connect to the patch panel.

Figure 9: Example Voice Mail or DEI Connection to MiVoice Office 250 Base Server



NOTE Prior to the 5000 Release v6.0, the optional external voice processing system would have been the Voice Processing Unit (VPU — unsupported since v3.0) or Enterprise Messaging (unsupported as of v6.0). Beginning with 5000 Release v6.0, NuPoint Unified Messaging (UM) is the supported external voice processing system.

Installing Modules in the DEI

Each DEI chassis attached to the Mitel controller chassis may be equipped with 16-port Digital Endpoint Modules (DEM-16) and/or eight-port Single Line Modules (SLM-8). Installation instructions for the DEM-16 and the SLM-8 follow, and an example of DEM-16 and SLM-8 circuit mapping appears in Figure 13 on **page 232**.

Installing a Digital Endpoint Module (DEM-16)

Each DEM-16 supports up to 16 digital telephones. Digital telephones include classic Inter-Tel-branded digital telephones, newer Mitel digital telephones, and Single-Line Adapters (SLA). This section provides the procedures you need to install the DEM-16. Procedures for installing digital telephones and SLAs appear in the following section.

Each DEI chassis accepts up to three DEM-16s to provide digital capability on the MiVoice Office 250. See the **Specifications** and **Product Description** chapters for additional module information.

NOTICE

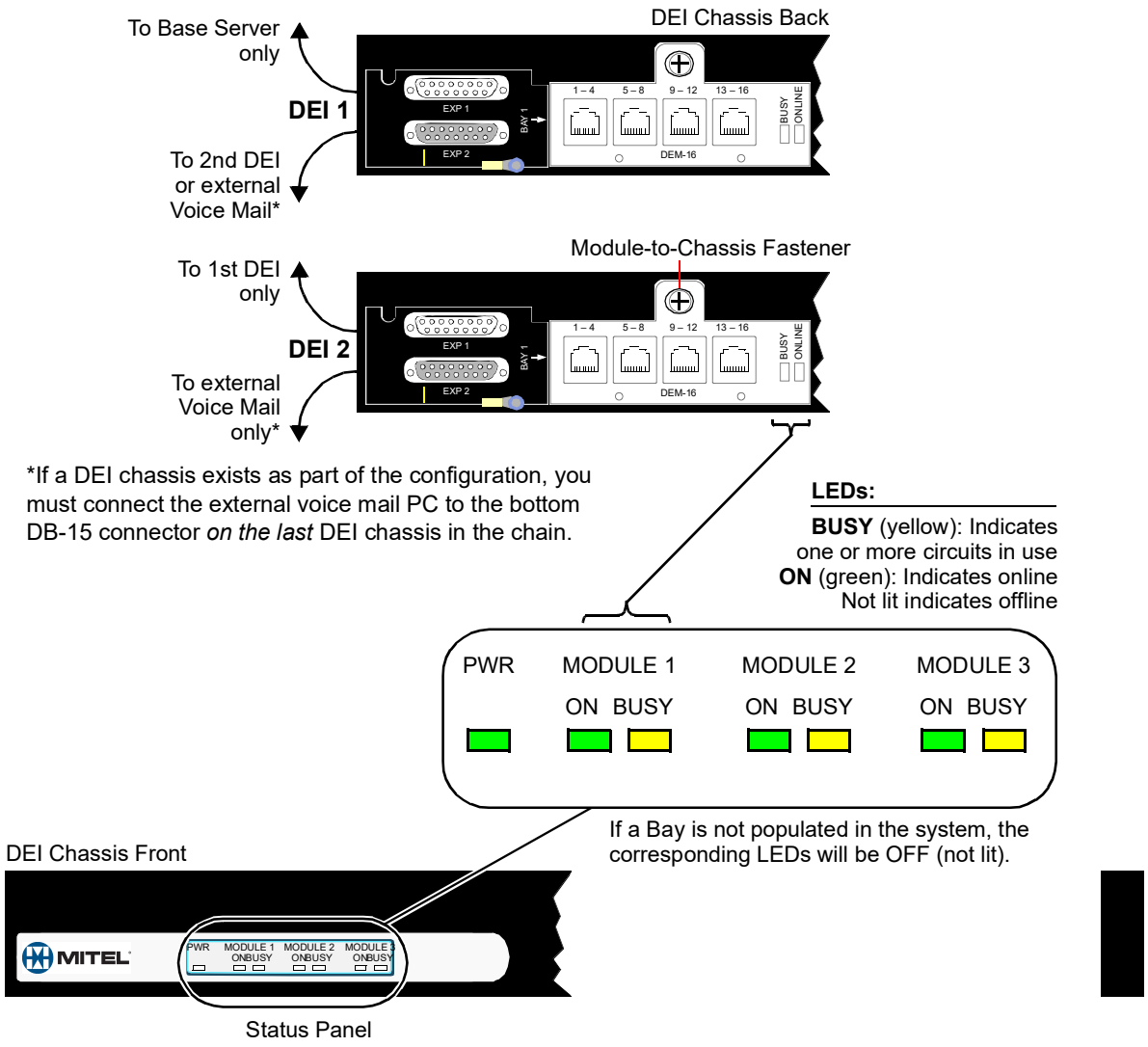
Electrostatic Discharge (ESD). MiVoice Office 250 modules and components are static-sensitive. When working with the modules, use a properly terminated anti-static wrist strap. Any static charge (no matter how small) must be discharged from the body before touching the modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will **not** be replaced.*

IMPORTANT	The DEM-16 can be removed with power on; however, any calls on those circuits will be dropped. DEM-16s can reside in any bay, but DB Programming will need to be changed if you move a module (that was previously programmed) into a different bay location.

To install a DEM-16:

1. Using a properly terminated, anti-static wrist strap, remove the module from the protective anti-static bag. See Figure 8 on **page 194** for a DEI chassis and DEM-16 example for the remaining steps in this procedure.
2. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops. Do ***not*** use force.
3. Tighten the fastener at the top of the module, securing the module to the chassis. Do ***not*** over-tighten this fastener.
4. Continue to “Verifying and Securing Cable Connections” on page 165.

Figure 10: DEM-16 in DEI 1 and DEI 2 Chassis



NOTICE

Equipment Damage Risk: Use adapters when connecting any 6-pin plug to an 8-pin jack. Although a 6-pin plug will fit into an 8-pin jack, the plastic on the 6-pin plug housing will damage pins 1 and 8 on the 8-pin jack. Not using adapters will increase the likelihood of failure (to pins 1 and 8 in the 8-pin jack) should that jack need to be used with an 8-pin plug in the future. Various adapter types are available based on how you choose to connect to the patch panel.

To remove a DEM-16:

1. Using a properly terminated, antistatic wrist strap, loosen the fastener at the top of the module (this fastener secures the module to the chassis).
2. When the **BUSY** LED is not lit (not green), grasp the module handle and carefully remove the DEM-16 from the chassis.

Installing an Eight-Port Single Line Module (SLM-8)

This section provides specifications, drawings, and procedures needed for installing an SLM-8, part number 580.2101.

Hardware

Table 39 summarizes the physical dimensions of the SLM-8 module.

Table 39: Eight-Port Single Line Module (SLM-8) Dimensions

	USA Customary	Metric
Length	8.375 in.	212.725 mm
Width	3.875 in.	098.425 mm
Height	1.000 in.	025.400 mm

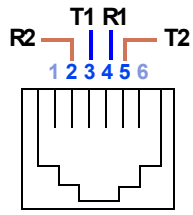
Each SLM-8 is equipped with four RJ-14 jacks, and each jack represents two single line ports, as shown below in **Figure 11**. For the applicable wiring diagram of SLM-8 ports, see Figure 12 on **page 231**. All SLM-8 ports support Caller ID transmitters.

Single Line Module Pinouts

The pinout design for four-port and eight-port Single Line Module jacks (SLM-4 and SLM-8) is the same. Each RJ-14 jack represents two single line ports for connecting single line analog phones. The middle four pins in the RJ-14 connector are wired in accordance with Uniform Service Order Code (USOC) T568A specifications, as shown in **Figure 11**.

Figure 11: RJ-14 Jack Pinout Diagram

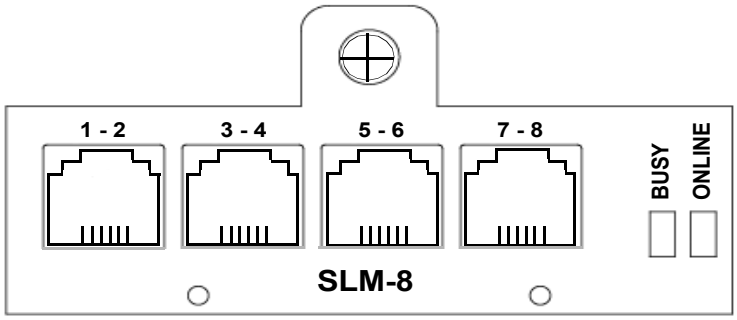
Each SLM-8 is equipped with two light-emitting diode (LED) status lamps, one yellow and labeled BUSY, and one green, labeled ONLINE. The yellow BUSY LED indicates that one of the ports programmed on the module is processing an active call. The green ONLINE LED indicates that the module is recognized by the system and ready for use.



T1–R1 = Port 1 Tip and Ring
T2–R2 = Port 2 Tip and Ring

IMPORTANT	All cabling connected to the RJ-14 interfaces should be constructed of standard 24-gauge twisted pair cable. To ensure reliable operation of single line phones connected to the MiVoice Office 250, do *not* use flat ribbon cabling or untwisted pair cabling.

Figure 12: Eight-Port Single Line Module (SLM-8), Front View



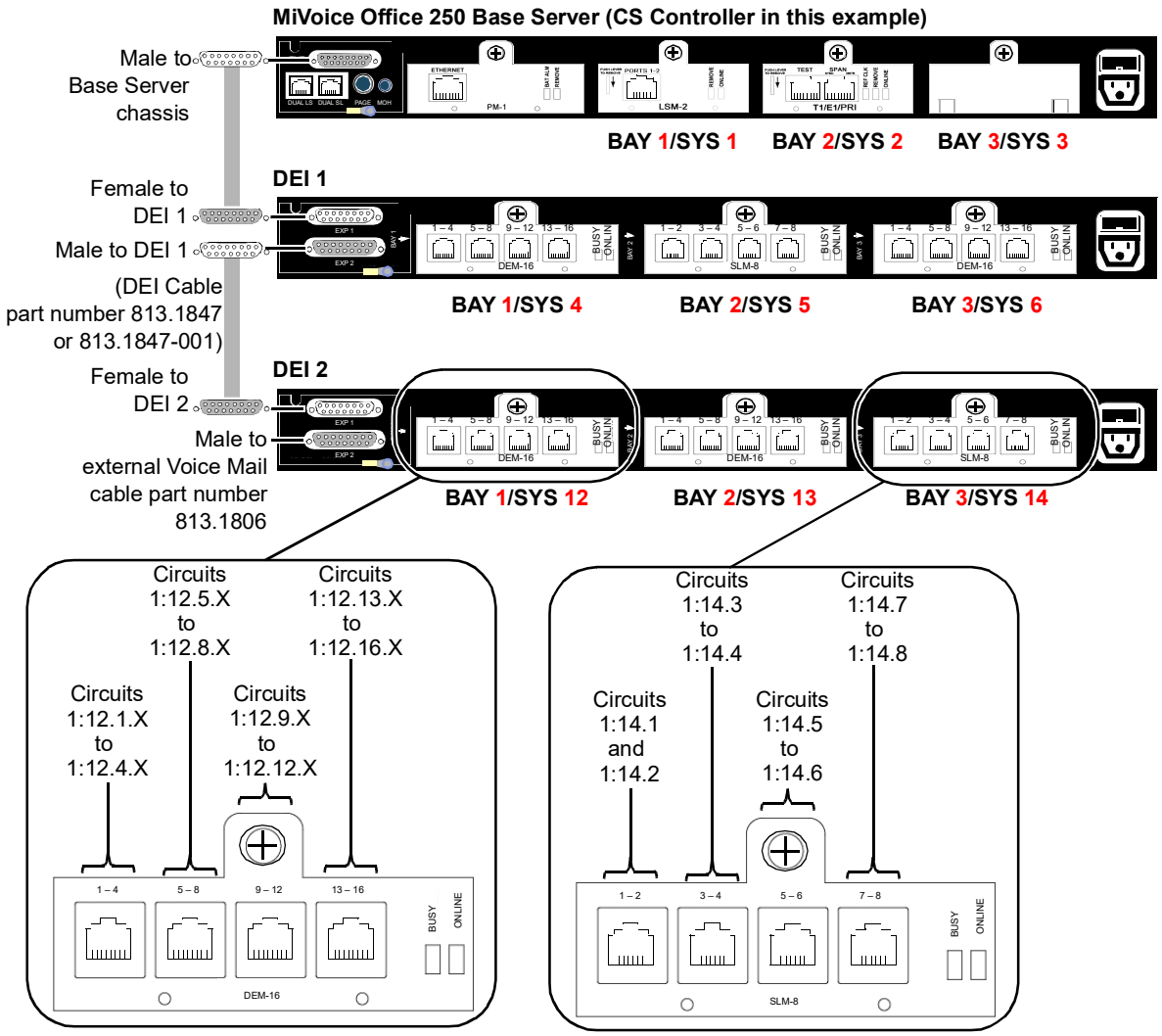
To install the SLM-8 module in a DEI chassis:

1. Put on ESD (electrostatic discharge)-protective gear and ground yourself.
2. In DB Programming, uninstall any module that may be in the bay where you want to install the SLM-8.
3. Remove the physical module in accordance with the applicable procedure—trunk modules require the REMOVE LED to be lit before the module can be safely removed.
4. Insert the SLM-8 module into the empty bay. The green LED labeled ONLINE illuminates, indicating that the new hardware has been detected and the Single Line (SL) application is running.
5. Using the IP address of the Mitel 5000, open a DB Programming session to the system through the System Administration & Diagnostics application.

NOTE The programming portion of the SLM-8 installation procedure may be performed in an offline session of DB Programming and then restored to the system using the Database Restore utility.

6. When DB Programming displays, in the left pane select System and then Communication Server.
7. Click the Uninstalled bay and click the down arrow.
8. Select **Single Line Module - 8**.
9. Program the SLM-8 module in the chosen uninstalled bay.
10. Click the SLM-8 folder and configure the ports.
11. Program the individual single line phones in DB Programming. For programming details, refer to the *MiVoice Office 250 Features and Programming Guide*.
12. Connect and test the phones.

Figure 13: SLM-8 and DEM-16 Circuit Mapping Example



LEGEND:

BAY 1/SYS 4 = Bay # in the chassis is the same as the equivalent System # in Message Print.

	Circuits		
	MiVoice Office 250 Base Chassis	DEI 1	DEI 2
Bay 1	1:01.1.X through 1:01.2.X	1:04.1.X through 1:04.16.X	1:12.1.X through 1:12.16.X
Bay 2	1:02.1.X through 1:02.30.X	1:05.1 through 1:05.8	1:13.1.X through 1:13.16.X
Bay 3	Bay not occupied	1:06.1.X through 1:06.16.X	1:14.1 through 1:14.8

In this example, a T1/E1/PRI module is configured for E1/PRI.

STATION CABLING CONNECTIONS

At this point in the installation, all cabling to the MDF, patch panel, optional rack, and stations should be completed. This includes all cables that terminate at the chassis modules.

To complete cabling:

1. Plug all applicable cables into their respective modules and chassis ports.
2. Connect the modules to MDF blocks or a patch panel using appropriate cabling. For digital station cabling, see the information beginning in the section below.

Connecting Station Cables to the MDF

This section describes connecting station cables to the MDF in a USA or European system.

Connecting DEM-16 Station Cables to a U.S. System

Digital telephones are connected to DEM-16 ports. Single line devices are connected to Single-Line Adapters (SLA) which are in turn connected to available DEM-16 circuits.

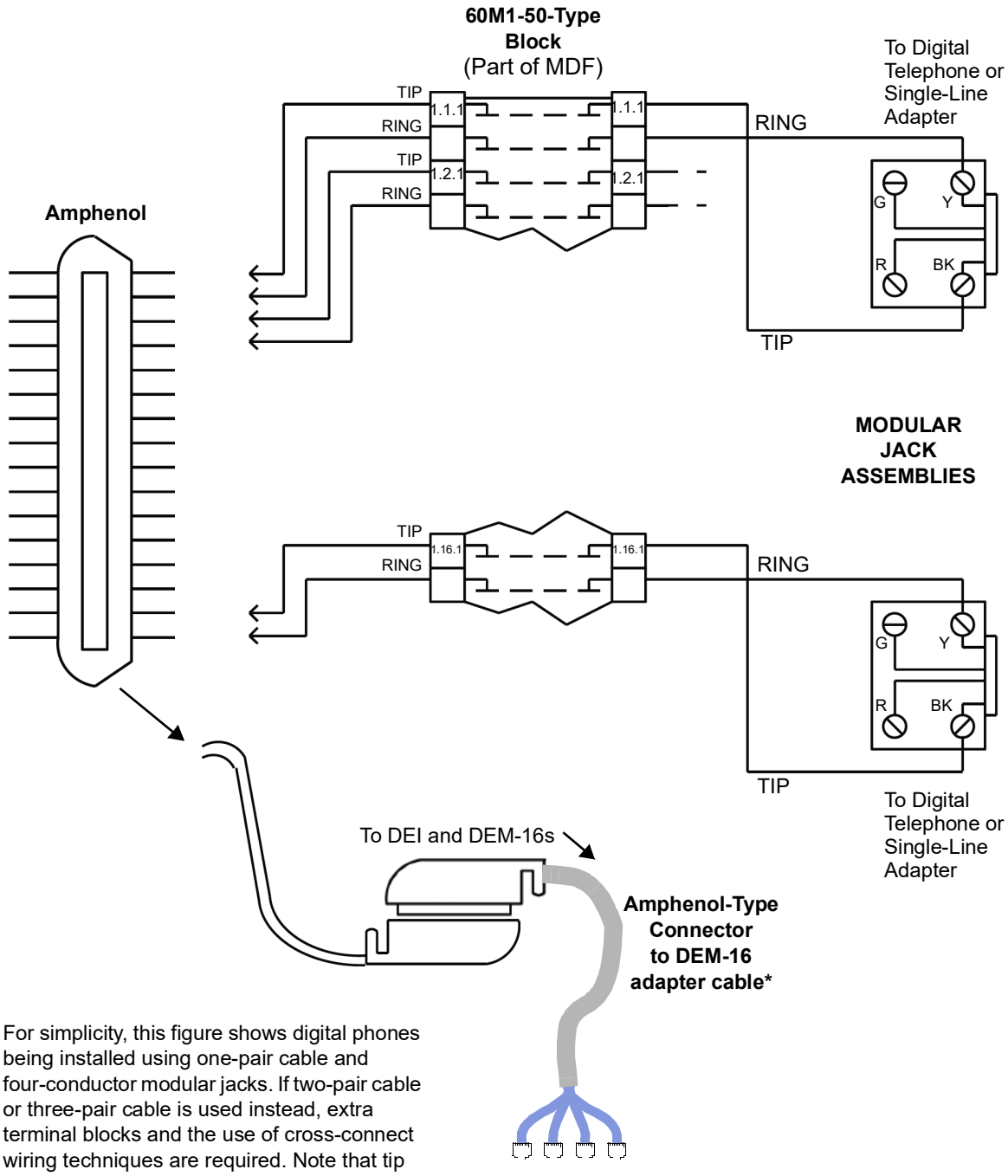
Besides single line phones, playback devices, and off-premises extensions, SLAs can also be used to install DID trunks. Any combination using the two circuits on the SLA is possible. DID trunk installation instructions appear in “Installing DID Trunks on an SLA (USA Only)” on **page 1-301**.

After the station cables are run, connect them and the 25-pair cables to the DEM-16 blocks on the MDF as follows:

1. Ensure that both ends of each station cable are labeled with the circuit number of the associated station instrument.
2. Terminate each station cable on the *right* side of the corresponding station block. See Figure 14 on **page 234** through Figure 18 on **page 240** for station cable terminations.

   For simplicity, the figures show station instruments being installed using one-pair cable and four-conductor modular jacks. If three-pair cable and six-conductor modular jacks are used instead, extra terminal blocks and the use of cross-connect wiring techniques are required.
3. Using enough 25-pair cable to run from each block to each DEM-16 in the equipment chassis, make the termination cables. Label each connector and cable end with the appropriate bay number (for example, Bay2). These connectors will be attached to the DEM-16s after they are installed.
4. *If necessary:* Using standard electrical tape, tape two ferrite split beads around each 25-pair cable just below the Amphenol-type connector (as shown in Figure 19 on **page 157**).
5. Ferrite beads may be necessary to help reduce electromagnetic interference (EMI) and radio frequency interference (RFI). The part number for a single split bead is 808.1195.
6. Terminate the other end of each 25-pair cable on the *left* side of the corresponding station block.
7. *Do **not*** attach bridging clips until the loop resistance tests have been performed, as described on **page 244**.

Figure 14: DEM-16 Block Cable Terminations for Phones on a U.S. System



For simplicity, this figure shows digital phones being installed using one-pair cable and four-conductor modular jacks. If two-pair cable or three-pair cable is used instead, extra terminal blocks and the use of cross-connect wiring techniques are required. Note that tip and ring are connected to the second pair of terminals (black and yellow) on the modular jack.

***Available Adapter Cable:**

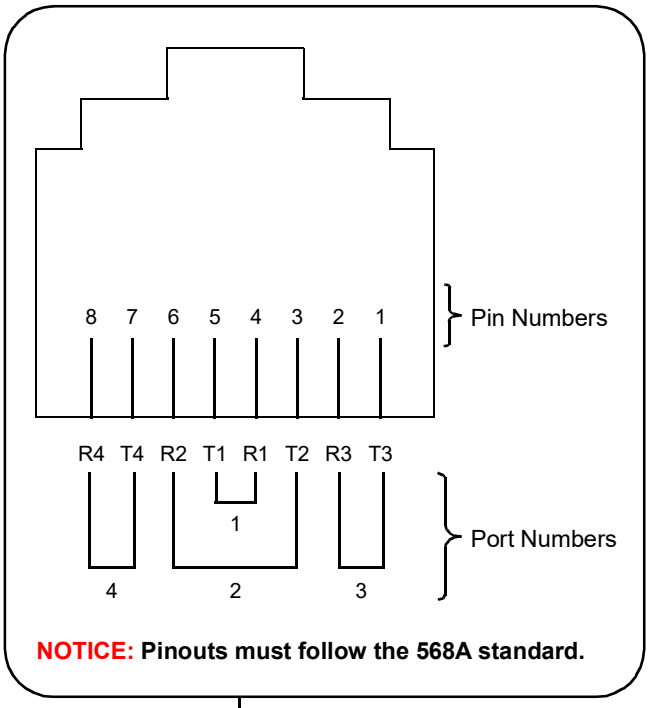
15 ft (4.5720 m) with Amphenol (part number 813.1814)

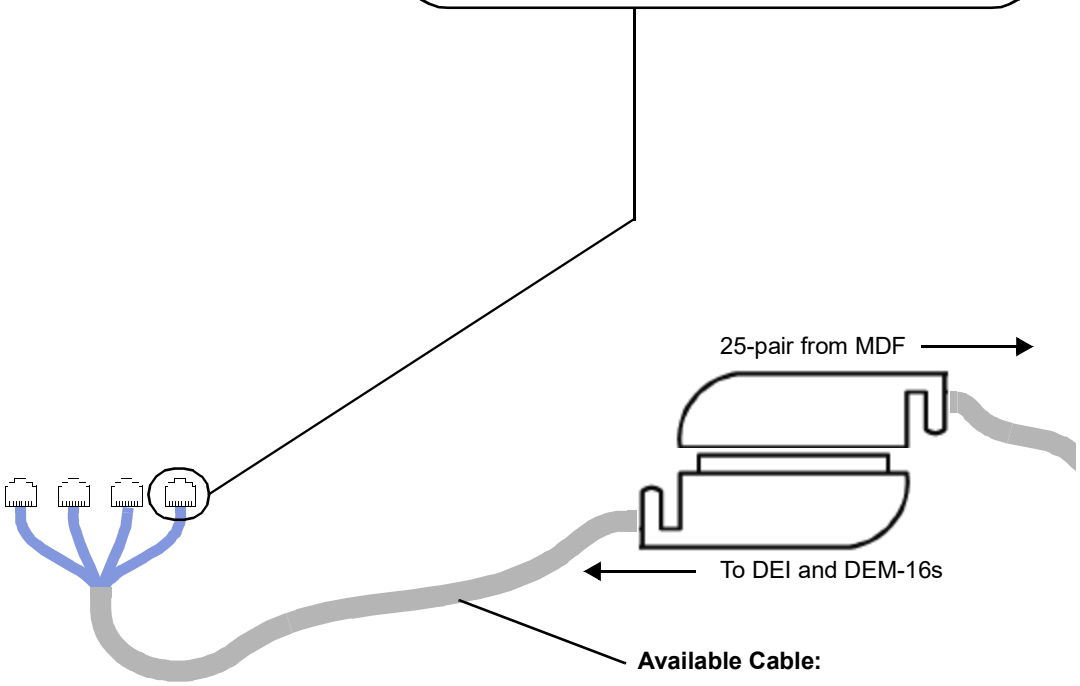
NOTE Unlike the 8520/8560 digital telephones, the newer 8528/8568 telephones use standard RJ11 wiring configuration as defined in TIA 968. See Table 22 on **page 112** for cable termination information.

Figure 15: DEM-16 Pinouts and Cabling

Port 1	Signal	Port 3	Signal
Pin-1	Tip 3	Pin 1	Tip 11
Pin-2	Ring 3	Pin 2	Ring 11
Pin-3	Tip 2	Pin-3	Tip 10
Pin-4	Ring 1	Pin-4	Ring 9
Pin-5	Tip 1	Pin-5	Tip 9
Pin-6	Ring 2	Pin-6	Ring 10
Pin-7	Tip 4	Pin-7	Tip 12
Pin-8	Ring 4	Pin-8	Ring 12

Port 2	Signal	Port 4	Signal
Pin-1	Tip 7	Pin-1	Tip 15
Pin-2	Ring7	Pin-2	Ring 15
Pin-3	Tip 6	Pin-3	Tip 14
Pin-4	Ring 5	Pin-4	Ring 13
Pin-5	Tip 5	Pin-5	Tip 13
Pin-6	Ring 6	Pin-6	Ring 14
Pin-7	Tip 8	Pin-7	Tip 16
Pin-8	Ring 8	Pin-8	Ring 16





Available Cable:

15 ft (4.5720 m) with Amphenol (part number 813.1814)

Table 40: Station Cable Terminations with a U.S. System

AMPHENOL Position #	Cable Pair	DEM-16 Bay 1	Circuit No.[a]	Extension No.[1]
26	W-BL	TIP	01.01.01	1000
1	BL-W	RING		
27	W-O	TIP	01.02.01	1001
2	O-W	RING		
28	W-G	TIP	01.03.01	1002
3	G-W	RING		
29	W-BR	TIP	01.04.01	1003
4	BR-W	RING		
30	W-S	TIP	01.05.01	1004
5	S-W	RING		
31	R-BL	TIP	01.06.01	1005
6	BL-R	RING		
32	R-O	TIP	01.07.01	1006
7	O-R	RING		
33	R-G	TIP	01.08.01	1007
8	G-R	RING		
34	R-BR	TIP	01.09.01	1008
9	BR-R	RING		
35	R-S	TIP	01.10.01	1009
10	S-R	RING		
36	BK-BL	TIP	01.11.01	1010
11	BL-BK	RING		
37	BK-O	TIP	01.12.01	1011
12	O-BK	RING		
38	BK-G	TIP	01.13.01	1012
13	G-BK	RING		
39	BK-BR	TIP	01.14.01	1013
14	BR-BK	RING		
40	BK-S	TIP	01.15.01	1014
15	S-BK	RING		
41	Y-BL	TIP	01.16.01	1015
16	BL-Y	RING		
42	Y-O	NOT USED		
17	O-Y	NOT USED		
43	Y-G	NOT USED		
18	G-Y	NOT USED		
44	Y-BR	NOT USED		
19	BR-Y	NOT USED		
45	Y-S	NOT USED		
20	S-Y	NOT USED		
46	V-BL	NOT USED		
21	BL-V	NOT USED		
47	V-O	NOT USED		
22	O-V	NOT USED		
48	V-G	NOT USED		
23	G-V	NOT USED		
49	V-BR	NOT USED		

Table 40: Station Cable Terminations with a U.S. System (Continued)

AMPHENOL Position #	Cable Pair	DEM-16 Bay 1	Circuit No.[a]	Extension No.[1]
24	BR-V	NOT USED		
50	V-S	NOT USED		
25	S-V	NOT USED		

a. The station circuit numbers shown above are for the first DEM-16 installed in the equipment chassis. (Bay 1). The extension numbers are the default numbers assigned to those circuits.

Connecting DEM-16 Station Cables to a European System

Each type of station instrument is connected to a specific port on a DEM-16. Digital telephones are connected to DEM-16s. Single line phones and playback devices are connected to either the built-in single line ports, or they can connect to Single-Line Adapters (SLA), which are in turn connected to available DEM-16 ports

After the cables are run, connect them to the station blocks on the MDF. Ensure both ends of each station cable are labeled with the circuit number of the associated station instrument. The station cables are pre-terminated on the upper half of the Krone block. The line plant wiring extension terminates on the lower half of the Krone block. See the following figures for station cable terminations:

- Figure 16 on **page 238**
- Figure 17 on **page 239**

Figure 16: DEM-16 Block Terminations for Phones on European System

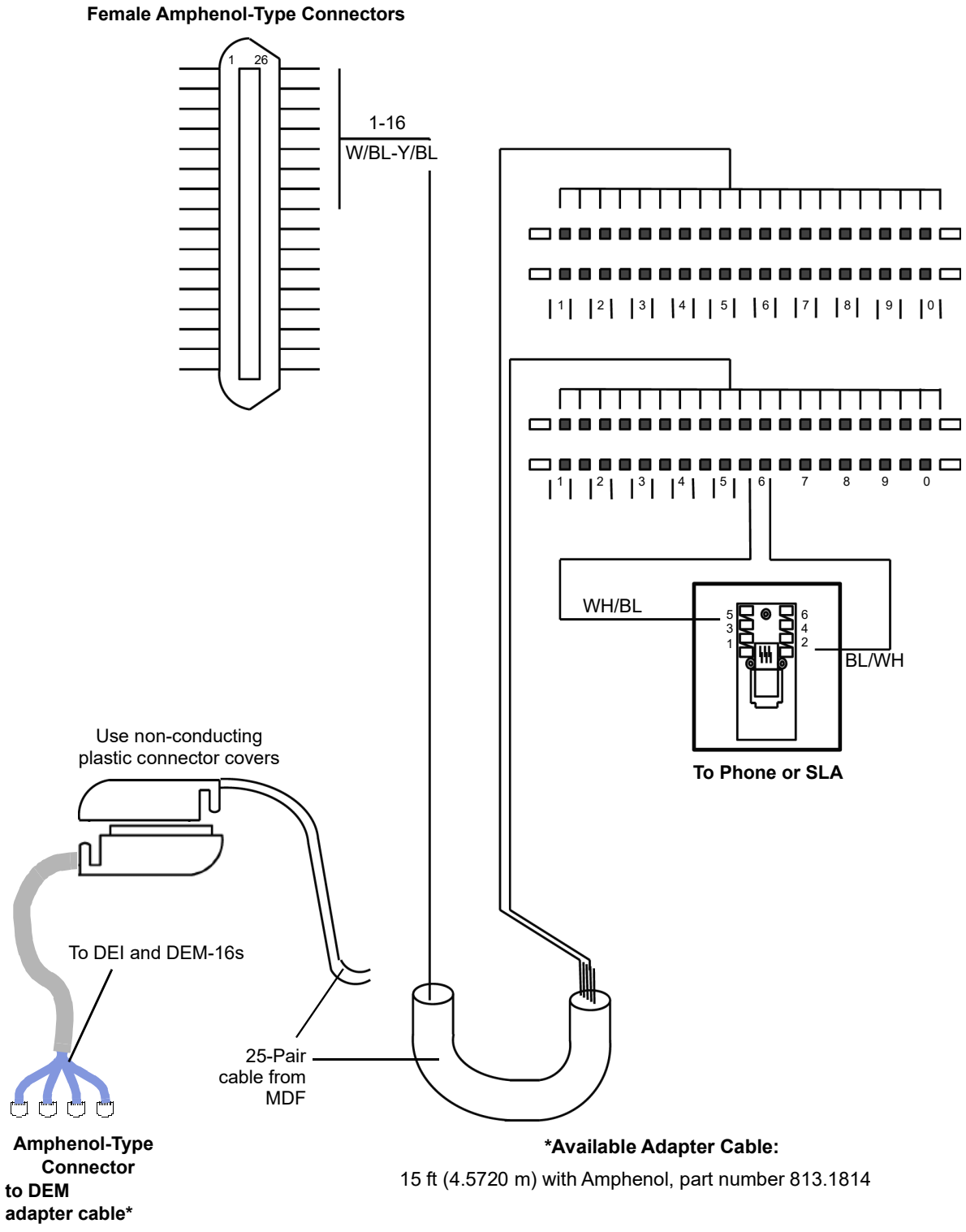
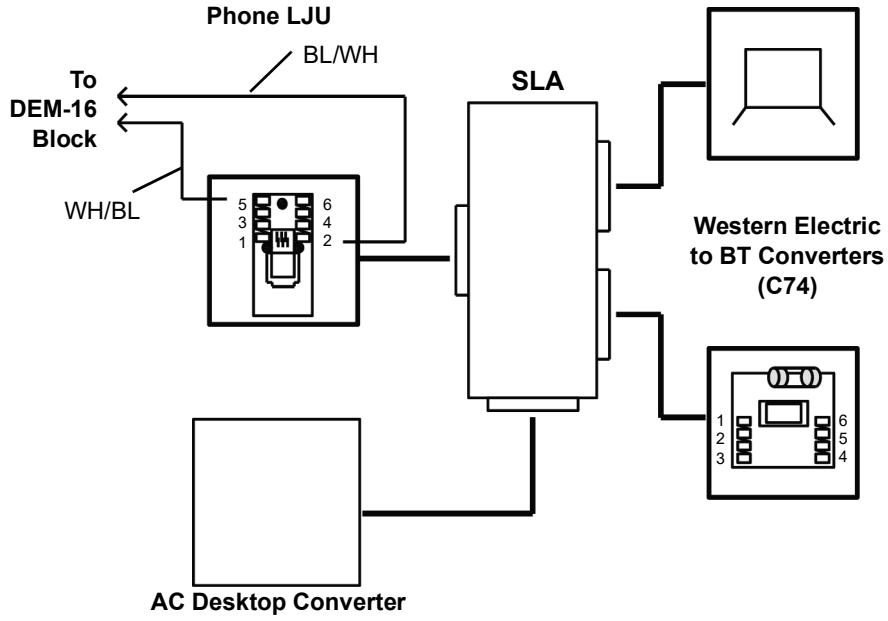


Figure 17: SLA Terminations for Single Line Phone with a European System



The SLA is supported with two Western Electric to BT converters (part number C74) for use when the attached devices are near the SLA. For longer range connections (up to 4.5 Km), an additional Krone block is required. This is connected to the SLA using a standard line cord— part number C27—and line jack unit (LJU).

Where exposed OPX cabling is used, also use BT14A/1 of BT21A surge arrestors on the OPX lines, and ensure the Krone field is earthed. Make the LJU at the far end of the cable a line master. This entire setup might be part of the main MDF or situated remotely, subject to chassis-to-SLA cable restrictions.

Figure 18: Off-Premises Connection [Europe Only]

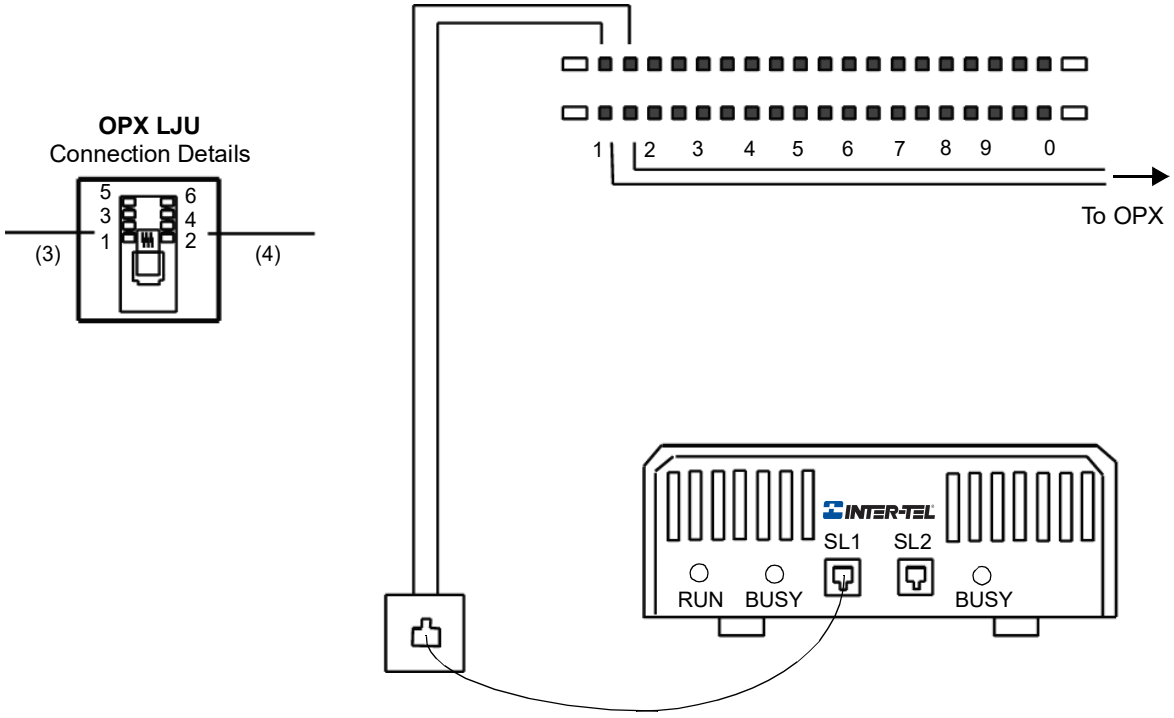


Table 41: Cable Terminations on MDF Blocks [Twin Block Cables] for DEM-16

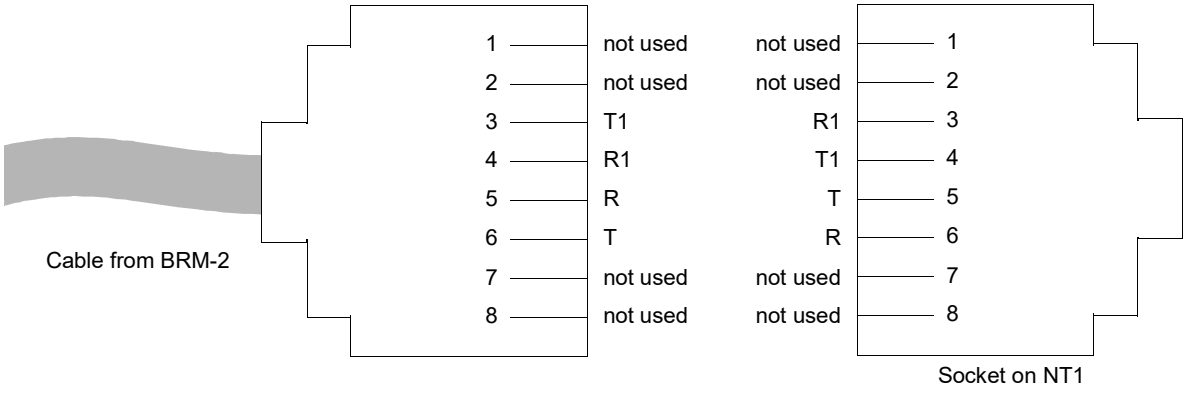
Cable Pair	Color	Plug Pin No.	BLOCK NO.	BLOCK PAIR	CIRCUIT
1	White/Blue Blue/White	1 26	1	1	01.01
2	White/Orange Orange/White	2 27		2	02.01
3	White/Green Green/White	3 28		3	03.01
4	White/Brown Brown/White	4 29		4	04.01
5	White/Gray Gray/White	5 30		5	05.01
6	Red/Blue Blue/Red	6 31		6	06.01
7	Red/Orange Orange/Red	7 32		7	07.01
8	Red/Green Green/Red	8 33		8	08.01
9	Red/Brown Brown/Red	9 34		9	09.01
10	Red/Gray Gray/Red	10 35		10	10.01

Table 41: Cable Terminations on MDF Blocks [Twin Block Cables] for DEM-16 (Continued)

Cable Pair	Color	Plug Pin No.	BLOCK NO.	BLOCK PAIR	CIRCUIT
11	Black/Blue Blue/Black	11 36	2	1	11.01
12	Black/Orange Orange/Black	12 37		2	12.01
13	Black/Green Green/Black	13 38		3	13.01
14	Black/Brown Brown/Black	14 39		4	14.01
15	Black/Gray Gray/Black	15 40		5	15.01
16	Yellow/Blue Blue/Yellow	16 41		6	16.01
17	Yellow/Orange Orange/Yellow	17 42		7	Do not use
18	Yellow/Green Green/Yellow	18 43		8	
19	Yellow/Brown Brown/Yellow	24 25		9	
20	Yellow/Gray Gray/Yellow	49 50		10	

Connecting BRM-2 Cables to a European System

Figure 19: BRM-2 Cable Connection for European System



Connecting DDM-16 Station Cables to the MDF

Digital telephones are connected to DDM-16 ports. Single line devices are connected to Single-Line Adapters (SLA) which are in turn connected to available DDM-16 circuits.

Besides single line phones, playback devices, and off-premises extensions, SLAs can also be used to install DID trunks. Any combination using the two circuits on the SLA is possible.

After the station cables are run, connect them and the 25-pair cables to the DDM-16 blocks on the MDF as follows:

1. Ensure that both ends of each station cable are labeled with the circuit number of the associated station instrument.
2. Terminate each station cable on the *right* side of the corresponding station block. See Figure 14 on **page 234** for an example of the station block, and see **Figure 20** below for an illustration of the special DDM-16 termination cable, part number 813.1910. Also see Table 42 on **page 242** for cable pinouts,
3. Using enough 25-pair cable to run from each block to each DEM-16 in the equipment chassis, make the termination cables. Label each connector and cable end with the appropriate bay number (for example, Bay2). These connectors will be attached to the DEM-16s after they are installed.
4. *If necessary:* Using standard electrical tape, tape two ferrite split beads around each 25-pair cable just below the Amphenol-type connector (as shown in Figure 19 on **page 157**).
5. Ferrite beads may be necessary to help reduce electromagnetic interference (EMI) and radio frequency interference (RFI). The part number for a single split bead is 808.1195.
6. Terminate the other end of each 25-pair cable on the *left* side of the corresponding station block.
7. *Do **not*** attach bridging clips until the loop resistance tests have been performed, as described on **page 244**.

Figure 20: DDM-16 Termination Cable

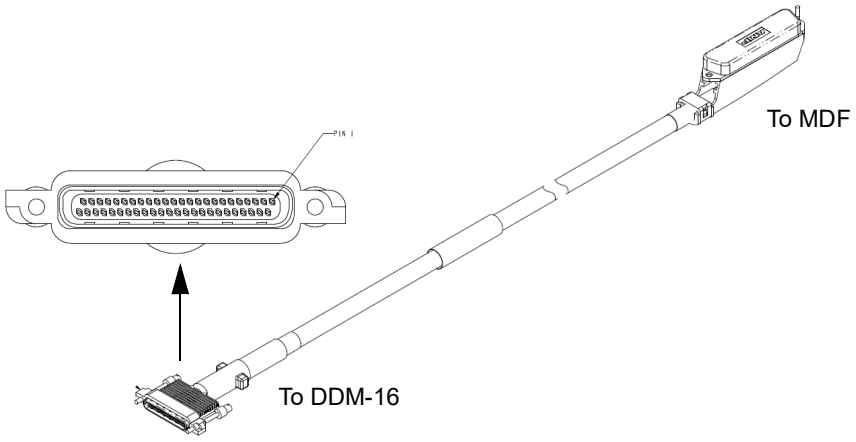


Table 42: DDM-16 Termination Cable Pinout

To DDM-16	To MDF	Signal Name	Color Pair
1	1	Ring 1	BLU-WHT
25	26	Tip 1	WHT-BLU
2	2	Ring 2	ORN-WHT
26	27	Tip 2	WHT-ORN
4	3	Ring 3	GRN-WHT
28	28	Tip 3	WHT-GRN
5	4	Ring 4	BRN-WHT
29	29	Tip 4	WHT-BRN
7	5	Ring 5	GRY-WHT
31	30	Tip 5	WHT-GRY

Table 42: DDM-16 Termination Cable Pinout

To DDM-16	To MDF	Signal Name	Color Pair
8	6	Ring 6	BLU-RED
32	31	Tip 6	RED-BLU
10	7	Ring 7	ORN-RED
34	32	Tip 7	RED-ORN
11	8	Ring 8	GRN-RED
35	33	Tip 8	RED-GRN
13	9	Ring 9	BRN-RED
37	34	Tip 9	RED-BRN
14	10	Ring 10	GRY-RED
38	35	Tip 10	RED-GRY
16	11	Ring 11	BLU-BLK
40	36	Tip 11	BLK-BLU
17	12	Ring 12	ORN-BLK
41	37	Tip 12	BLK-ORN
19	13	Ring 13	GRN-BLK
43	38	Tip 13	BLK-GRN
20	14	Ring 14	BRN-BLK
44	39	Tip 14	BLK-BRN
22	15	Ring 15	GRY-BLK
46	40	Tip 15	BLK-GRY
23	16	Ring 16	BLU-YEL
47	41	Tip 16	YEL-BLU

Station Loop Resistance Test

Do not use shielded cable. Achieve longer loop lengths by using heavier gauge cable, such as 22AWG. (For additional station cabling requirements and loop limits, see page 152.)

Perform the loop resistance test for each station cable individually.

1. *With a U.S. system only*, ensure that bridging clips have not been installed on the station blocks and that the station instrument is not connected to the modular jack assembly.
2. *With a U.S. system*, place a short across the BLACK/YELLOW wires (digital telephones) or RED/GREEN wires (single line phones) on the modular jack assembly.

   With a European system, place a short across the WHITE/BLUE wires (digital telephones) or BLUE/WHITE wires (single line phones) on the line jack unit assembly.
3. *With a U.S. system*, at the MDF, measure the resistance across the TIP and RING wires on the right (station) side of the station block.

   With a European system, at the MDF, measure the resistance across the WHITE/BLUE wires on the lower (station) side of the station block.

   The reading should not exceed the limits listed in **Table 43**. The resistance values (in ohms) indicate loop measurements; the distance values (in feet/meters) indicate the maximum one-way measurements from the equipment chassis. If necessary, longer loop lengths can be achieved by using a heavier gauge cable, such as 22 AWG (22 gauge thickness or larger—smaller number gauge = larger, or thicker, diameter).

   IMPORTANT Excessive and/or improperly made connections increase the resistance of a cable, which reduces the allowable cable run length.
4. Remove the short after the test is complete.
5. Repeat this test for each station cable.
6. With a U.S. system only, install bridging clips on the station blocks to complete the cable connections.

Table 43: Loop Limits for Digital and Single Line Phones and Equipment

Phones	Port Impedance	Loop Limit (24 AWG)
From Base Server chassis: • To a single line phone or playback device connected to the built-in port • To a single line phone or playback device driven by a SLM-4 port **From DEI chassis:** To a single line phone or playback device driven by a SLM-8	600 Ω	2000 ft (609 m) REN 3.0
From **DDM-16 (HX only)** or DEI chassis: • To digital telephones • To a digital telephone equipped with PCDPM and MDPM and connected to PKM/DSSs or SLA	62 Ω	1200 ft (366 m)
From an MDPM: To a single line phone	195 Ω	3800 ft (1158 m)
From an SLA: To single line phones or playback devices. Supports facility code OL13C.	800 Ω	15,560 ft (4742 m)

Connecting Off-Premises Extensions

Off-premises extensions (OPX) are remote single line DTMF stations connected to the MiVoice Office 250 through telephone company [network provider] OPX circuits or customer-provided circuits. The circuits are connected to Single-Line Adapter (SLA) circuits, which are connected to available DEM-16 circuits or to T1M or T1M-2 module circuits.

Introduction

SLAs do *not* support visual message waiting applications. If single line phones equipped with message waiting lamps are installed on SLAs, the lamps will *not* function. However, single line phones can be configured in DB Programming to receive audible message waiting indication tones.

OPX devices must be AC ringer-equipped. The SLA onboard high-voltage ringing adapter (HVRA) provides isolation from the telephone company OPX circuits as required by FCC regulations.

Each SLA can support one or two OPX stations. SLA circuits support AC-ringing single line phones, fax machines, playback devices, etc.

With some OPX installations, voice volume levels may be degraded due to constraints on the voice-frequency range inherent in the telco [network provider] or customer-provided circuit. To increase the voice-frequency range, Mitel recommends installing a two-wire, negative impedance repeater unit. For details, see the **Specifications** chapter.

The repeater unit is installed at the MDF between the SLA and the cabling to the OPX station. For proper installation with the MiVoice Office 250, make the following connections to the appropriate pins on the repeater unit:

- Battery (–24 to –56 VDC)
- Ground
- System Tip
- System Ring
- OPX Tip
- OPX Ring

For identification of correct pin locations and for information about operating and adjusting the repeater unit, refer to the manufacturer's instructions.

Installing an OPX Station

To install the OPX stations:

1. At the MDF, terminate each telco [network provider] OPX circuit or customer-provided circuit on the desired SLA circuit. See Figure 4 on **page 303**.
2. At the OPX location, terminate each telco [network provider] OPX circuit or customer-provided circuit on the appropriate single line phone modular jack assembly [line jack unit].
3. Program the station and/or trunk as appropriate in DB Programming.

4. Test for quality.

MAINTENANCE TASKS

The following sections provide procedures for performing tasks that are required multiple times when installing, upgrading, expanding, or servicing a MiVoice Office 250.

Powering Up the System

NOTICES

Do ***not*** plug in the UPS power cord unless you are ready to boot the system. There is no power-on switch on the Base Server chassis. Plugging in the power cord begins the boot process. If power to the system fails (prevented by using a UPS), then comes back, the system boots automatically. Under normal circumstances, as long as the power cord is plugged into the chassis, the system is up and running. If the system is shut down using the LCD panel, *use the center button next to the LCD panel to power up the system*.

Possible Memory Card Corruption: By adhering to the following precautions you can help prevent corruption of data on the compact flash-type memory card:

- Do ***not*** remove or install the memory card while the system is up and running or power is otherwise supplied to the Mitel 5000, as this may damage the memory card. Shut down the system using the LCD panel and unplug the power cord before removing or inserting the memory card.
- Using any compact flash-type memory card other than one provided by Mitel is ***not*** supported.
- Do ***not*** place the memory card into any other PC. Doing so may corrupt the file system and software.

To power up the system:

1. Make sure all chassis modules are installed and fastened to the chassis.
2. Make sure the chassis is properly grounded and check all connections for loose, cracked, or broken components.
3. Make sure all cabling is securely attached to the modules, or any other components, at the back of the chassis. At times, cables can loosen and drop a connection.
4. Make sure the compact flash-type memory card and USB security key (if needed) are installed.
5. Plug in the chassis power cord into the UPS device. Plug in the UPS power cord. The system boots and is ready to begin initial configuration and programming.
6. If this is an upgrade, do the following:
   a. Perform a database backup.
   b. Start the Administrator Web Session interface and upgrade the system software.
7. Install the current DB Programming version.
   a. Upload the current license file, as applicable to your site.
   b. Disable Unified Voice Messaging (UVM) in DB Programming if NuPoint UM only will be used as the voice processor. Note that UVM can be used in conjunction with NuPoint UM on the same system. From the DB Programming menu bar, select **Operations** –> **Voice Processing Operations** –> **Disable Unified Voice Messaging**.
8. Plug in the remaining power cords in this order:
   a. Plug in the first DEI chassis.
   b. Plug in the second DEI chassis.

c. Plug in the external voice mail system, if equipped. For information on attaching an external voice processing system, see page 333.

9. If applicable (first-time installation), follow the initial system startup in accordance with the section titled, “Setting Initial System Configuration” on **page 1-166**. Otherwise, continue to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*.

Powering Down the System

To power down the system:

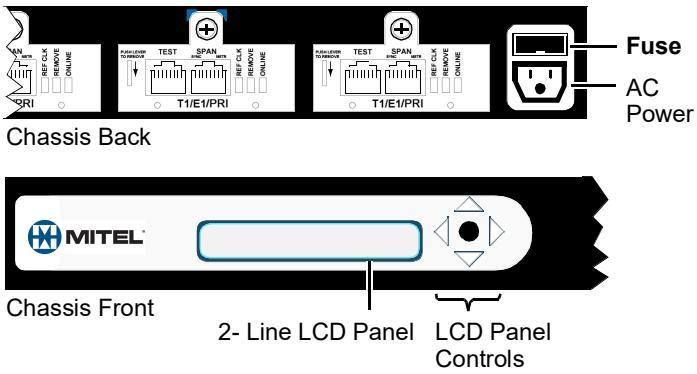
1. Make sure Database (DB) Programming is not running. If DB Programming is running when you make programming changes using the LCD panel, the changes will not be updated until the system is restarted.
2. From the LCD panel, press the down arrow twice to navigate to the **System** menu.
3. Press the center button once to access the System menu, then press the down arrow until you get to the **Shutdown** menu.
4. Press the center button once to access the Shutdown menu.
5. At the prompt, press the down arrow once to select the **Yes** option.
6. Press the center button to execute the **Yes** option. The system reboots.

NOTE If the system does not power down, do ***not*** unplug the power cord. If the system hangs up, you can force the power off by pressing—all at the same time—the center LCD panel button and the right arrow and left arrow buttons.

Replacing the Power Fuse

The CS Controller has a replaceable power fuse, part number 802.2028. The fuse is located above the power cord connection on the back of the chassis, as shown in **Figure 4**.

Figure 4: Base Server Chassis Power Fuse



To replace the power fuse:

1. Perform a Database backup—for details, refer to the *MiVoice Office 250 Features and Programming Guide*. You can also back up the database using the LCD panel—see page 169.
2. Perform a voice processing database backup. With a Base Server-only system running UVM, Mitel recommends using a USB flash drive attached to the USB-A port on the front of the CS Controller chassis. Do *not* back up VM to the compact flash-type memory card. For the backup procedure, refer to the *MiVoice Office 250 Features and Programming Guide*. For a list of USB flash drives that have been tested as compatible with the system, see “USB Flash Drives” on **page 1-83**.
3. Shut down the system using the LCD panel. See Figure 4 on **page 169** for an illustration of the menu locations.
4. Unplug the power cord.
5. Remove the fuse from above the power connection at the back of the chassis, then replace the fuse.
6. Make sure the fuse is seated correctly, then connect the power cord. The system powers up and begins the boot process.

Replacing the Processor Module Battery

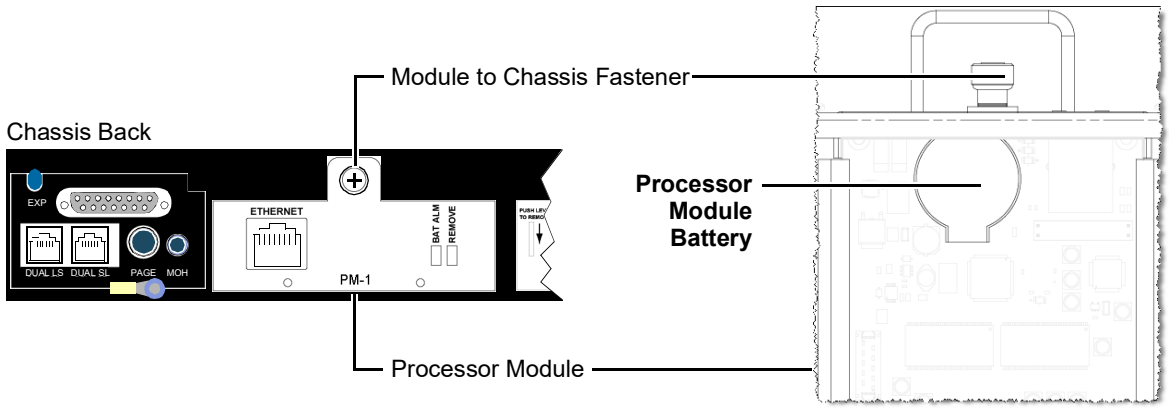
Because the CS Controller's PM-1 battery, part number 815.1026, is located on the module, do *not* remove the cover to replace the battery.

NOTICES

Equipment Damage. Danger of explosion if battery is incorrectly replaced. Replace only with the same or equivalent type recommended by the manufacturer. Dispose of used batteries according to the manufacturer instructions.

Electrostatic Discharge (ESD). MiVoice Office 250 modules and components are static-sensitive. When working with the modules, use a properly terminated anti-static wrist strap and cover the work surface with the bag that protected the module during shipping, or place the component on some other anti-static surface. Any static charge, no matter how small, must be discharged from the body before touching modules or other components. *The warranty for this equipment does not cover damage caused by static or mishandling. Modules or components damaged in such a manner will not be replaced.*

Figure 5: Processor Module Battery



To replace the PM-1 battery:

1. Perform a Database backup using DB Programming. (You can also back up the database via the LCD panel.) Refer to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*, for details.
2. Perform a voice mail backup. Use a USB flash drive attached to the USB-A port on the front of the Mitel 5000 chassis. Do *not* back up VM to the compact flash-type memory card.
3. Shut down the system using the LCD panel. See Figure 4 on **page 169** for an illustration of the menu locations.
4. Unplug the power cord. Make sure you are wearing a properly terminated, ant-static wrist strap for the remainder of this procedure.
5. Loosen the fastener at the top of the module.
6. Remove the PM-1 from the CS Controller chassis and place it onto the anti-static bag that came with the module, or onto some other anti-static surface.

7. Locate the battery and carefully remove it from the PM-1. Replace the battery with the new battery.
8. Insert the module carefully so that it aligns with the rails inside the chassis, then gently slide the module into the chassis until it stops. Do not use force.
9. Finger-tighten the fastener at the top of the module, securing it to the chassis. Do ***not*** over-tighten this fastener.

Installing the USB-B Port Serial Driver

The USB-B port on the front of the Mitel 5000 chassis allows you to connect a laptop and view online messages. However, a driver must be installed on the laptop to allow the laptop to use this USB-B port. For information on viewing online messages, see “Using Terminal Emulation to Retrieve Online Messages” on **page 1-252**.

After installing the system software, the latest USB drivers (USBDriverWin7 and USBDriverWinXPVista) are available in the Drivers folder within the installation folder (for example, C:\Program Files\Mitel\5000\Drivers). The new drivers support the Windows XP, Windows Vista, and Windows 7, 8, and 10 operating systems. To install a driver, run the appropriate executable file. If upgrading from a previous DB Programming installation, the previously installed driver executable and readme files will remain in the Drivers folder.

NOTE When using the USB-B port, use a cable designed for the USB 2.0 specification.

To install the USB-B serial driver:

1. Do not connect any cable into the USB-B port until instructed.
2. Locate the Driver folder on the PC where you installed DB Programming (e.g., `C:\Program Files\Mitel\5000\Drivers`. This folder contains the zip file that contains the driver installation files and driver components.
3. Double-click the appropriate file to install the driver files. Follow the Wizard screens.
4. Connect a USB 2.0 cable from a laptop to the USB-B port. The Found New Hardware Wizard appears.
5. Point the New Hardware Wizard to the file type it requests, to complete the driver installation.

Retrieving SMDR or Error Message Output Data

SMDR/Error message output is provided over an IP connection. You can retrieve it through an external voice processing system using a specific IP address and port number. This feature is not available through Unified Voice Messaging. SMDR can be buffered to System Manager, if enabled or connected.

For information on retrieving SMDR/Error Message information, refer to the Diagnostics and Troubleshooting chapter in the *MiVoice Office 250 Features and Programming Guide*.

Using Terminal Emulation to Retrieve Online Messages

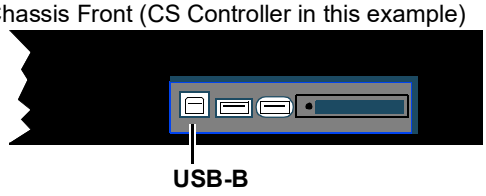
The USB-B port is provided to allow monitoring information online using a terminal emulation program such as Windows® HyperTerminal. You will need a USB cable with a standard USB Type-A connector on one end and a Type-B connector on the other end. This interface is only supported for terminal emulation use. Do not plug any other device (USB hub, etc.) into the USB-B connection. Also, to avoid a drop in system performance, do not monitor a live system for any length of time other than for troubleshooting purposes.

NOTE HyperTerminal does not support SSH encryption, which is required for TCP/IP connections. The PuTTY® application can be used for these types of connections. A copy of PuTTY is generally included on system software DVDs.

To retrieve online message information:

1. Pick a time to monitor system information when the system is not in use or at a time when system traffic is the lowest, if possible.
2. Make sure you have installed the driver that will allow a PC or laptop to access the USB-B port (“Installing the USB-B Port Serial Driver” on page 251).
3. Attach a USB 2.0 cable to the USB-B port on the front of the chassis.
4. Attach the other end of the cable to a PC or laptop USB port (standard USB connector).
5. Start the terminal emulation program and set any configuration options the program needs to communicate with the MiVoice Office 250. See the documentation that came with the terminal emulation program for this information. For the terminal emulation program you use, set the following options if available:

   Chassis Front (CS Controller in this example)

   - Communications Port: **COM3**
   - Bits Per Second: **115200**
   - Data Bits: **8**
   - Parity: **None** or **n.**
   - Stop Bits: **1**
   - Flow Control: **None**.
6. Enter the appropriate administrator account username and password. The “OLM>” prompt appears on the screen, indicating you have logged in to the Online Monitor (OLM) shell.

   To view a list of available commands, type **help** or **?** at the OLM prompt, then press the **ENTER** key.

   To exit, type **EXIT** or **.** (period), then press the **ENTER** key.

Removing System Voice Prompts and Phone Firmware

IMPORTANT	Perform a Backup Database Save after setting the TFTP Pathname and TFTP IP Address fields. Since the upgrade could restart call processing (CP), the TFTP address/path change will most likely be lost when CP restarts. When CP restarts with a new version it does not use the database stored in persistent memory, instead it uses the backup database and does any conversions necessary for the new version. By performing a backup database save, all recent changes including the TFTP address/path will be preserved.

If a phone firmware or voice prompt package is added to the system, and later it is determined that you need to revert back to a previous version of the same package, the package will need to be removed and the previous version installed.

NOTICE
Upgrading system software using the Update All option requires a system reboot. Do ***not*** use this option during normal business hours.

IMPORTANT	Before removing any packages, make sure you have the prior version of that package so that it can be reinstalled, if needed.

Do the following:

1. Launch and log on to AWP.
2. Click the **System Management** tab, and then click the **System Upgrade** link.
3. Click the package name you want to remove.
4. Click **Remove** to remove the selected package.
5. Install the previous version of the package you removed.
6. Reboot the system, if prompted. If you chose the Update All option, the system will reboot, if needed.

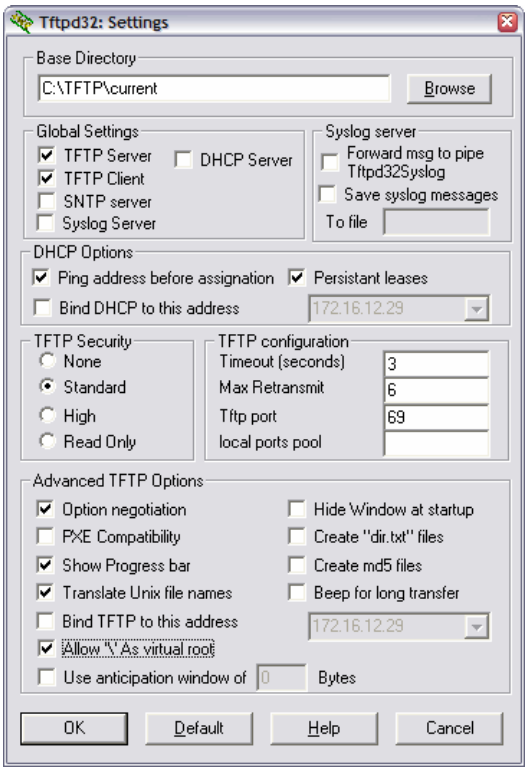
TFTP Server Configuration Example

This example provides information on some configuration settings for the Tftp32d TFTP server (http://perso.wanadoo.fr/philippe.jounin/tftpd32.html). Depending on the TFTP server you are using, these settings may be located in other dialog boxes or locations. Mitel does not provide support for this TFTP server. It is shown here as an example only. For DB Programming information discussed in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*.

To set TFTP server properties:

1. Launch the TFTP server application and click the **Settings** button.

Figure 6: Example TFTP Server Configuration



2. Configure the following settings. An example is shown on the previous page.
   - **Global Settings**: **TFTP Server** and **TFTP Client**.
   - **DHCP Options**: **Ping address before assignation** and **Persistent leases**.
   - **TFTP Security**: Standard
   - **TFTP configuration**: Leave at default settings.
   - **Advanced TFTP Options**:
     - Option negotiation
     - **Show Progress Bar**: Displays file transfer information.
     - **Translate UNIX file names**: Without this checked, the operation will fail.
     - Allow '\' As virtual root
3. Click **OK**, then restart the TFTP server application. The TFTP server must be running before you attempt to upgrade the MiVoice Office 250.

Upgrading Compact Flash-Type Memory

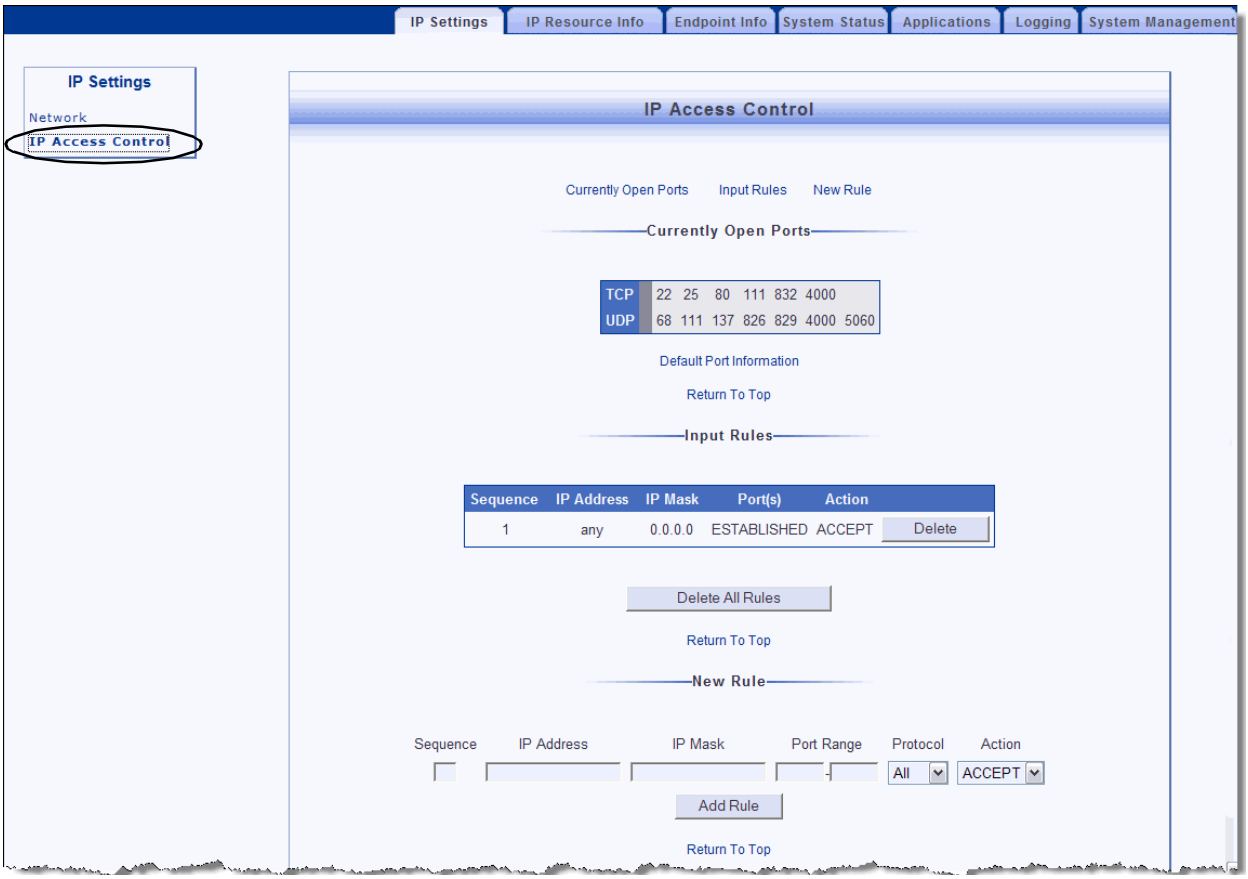
NOTICE

Version 3.x requires a minimum 512 MB compact flash-type memory card. Version 4.0 or later requires a minimum one GB compact flash-type memory card. The 512 MB card is not supported with v4.0 or later.

For additional information about Unified Voice Messaging storage capacities, see page 72. For database and voice processing data information, refer to the *MiVoice Office 250 Features and Programming Guide*.

To upgrade the compact flash-type memory card:

1. Schedule this upgrade at a time that will have the least impact on the customer.
2. Access the Administrative Web Portal pages and do the following:
   a. Click the **Advanced View** link at the top of the window, and then navigate to the IP Settings tab. Click the **IP Access Control** link in the navigation bar.
   b. Print the AWP page displaying this information. If any rules are listed here, you will need to recreate them after installing the new memory card. An example IP Access Control page appears on the next page. (See the notice that follows.)

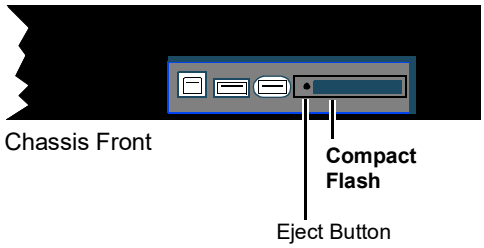


NOTICE

System Inoperability. Providing rules in the wrong sequence or constructed of the wrong content, could deny you access to the system through the Administrative Web Portal or through DB Programming.

3. Perform a database backup.with voice data included.
4. Power down the system and unplug the power cord. For details, see page 247.

5. Remove the memory card and insert the new memory card into the slot labeled COMPACT FLASH, as shown in the following CS Controller example.



6. Plug in the power cord.
7. Restore the database with voive data included.
8. Check for any software updates and install them, if necessary. Although the larger compact flash-type card has all the software included, always check for the latest software updates to the system at Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome). You will be prompted to reboot if a reboot is needed. If using the Upgrade All option, the system will reboot automatically, if needed.

NOTE Starting with the v4.0 software release, North American compact flash-type cards include both English and Canadian French language prompts. If additional language prompts are required, download and install the desired language packages.

9. Reload the license file.
10. Access the Administrator Web Session pages and do the following:
    a. Click the **Advanced View** link at the top of the window, and then navigate to the IP Settings tab. Click the **IP Access Control** link in the navigation bar.
    b. Enter any rules that were listed on this page prior to the upgrade.

PHONES

This section provides procedures for installing IP, digital, and analog single line phones that operate on the MiVoice Office 250. For phone illustrations, see “Phones“ on **page 479**. For end user feature descriptions, refer to the End User Features chapter in the *MiVoice Office 250 Features and Programming Guide*.

Important Notices

Please read this section before continuing.

NOTICES

- **Possible Functionality Degradation**. Mitel strongly recommends using the IP devices on a managed private network. If connected to the public Internet, IP devices will function, but the quality may suffer due to the dynamic bandwidth availability. The possible problems could be voice quality degradation, garbled speech, dropped calls, equipment resets, etc. Also, the VoIP suitability of any Internet connection can change at any time, with no advance notice. **Mitel cannot guarantee any voice quality when connected to the public Internet. Therefore, Mitel is not responsible for network quality issues that are caused by using the public Internet.** For recommended guidelines on IP network specifications, see page 123.
- **Possible Network Instability**. Do ***not*** connect the IP phone PC ports to each other, and do *not* connect them to a hub. Doing so may cause adverse network conditions. Also, Mitel recommends that you *not* connect additional hard IP phones to the PC ports. Due to the QoS (Quality of Service) of the hard IP phones, those located downstream will have a lower priority when sending data to the network. Under heavy network conditions, voice quality may be diminished.
- **Equipment Damage Hazard**. Only use a single appropriate power adapter with the IP device. Do not connect a U.S. power supply and a Universal power supply to the same IP device. Also, when using the Universal Power Supply with a barrel connector, no devices, such as hubs, should be inserted between the KS/SLA jack on the adapter and the LAN jack on the IP phone because power is supplied through the cable. To reduce the effects of AC voltage surges and spikes that may cause damage to the electronic components, a surge/spike protector with the following specifications is recommended:
  - Clamp voltage transients at 300 VAC within 5 nsec when exposed to waveforms as described in the ANSI/IEEE Standard C62.41-1980 (IEEE 587).
  - RFI/EMI noise reduction by at least 20 dB at frequencies between 5 kHz and 30 MHz.

  The Mitel DataNet CommSource division sells a surge/spike protector, part number 900.0500, that possesses all of the recommended specifications.

Installing Mitel 52xx/53xx IP Phones

Installation guides are packed with each Mitel 52xx/53xx IP phone (5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, and 5360). Refer to the instructions in these guides and to the information in this section. For specifications, see page 102.

This information is also available in PDF format on the Mitel eDocs Web site (http://edocs.mitel.com/).

Special Considerations for 52xx/53xx Phones

Consider the following before installing the Mitel 52xx/53xx phones:

- An IEEE 802.3af LAN-powered jack provides power to the phone using a LAN cable. The phone receives ethernet signals and power through the same cable, also known as Power over Ethernet (PoE).
- An Ethernet power adapter provides power to the phone using a LAN cable. The Ethernet power adapter is connected to the LAN using a separate LAN cable connected to a jack. (The phone receives ethernet signals and power through the same cable.)
- The Mitel 5212 and 5224 are ***not*** supported for remote/teleworker locations.
- When using a Mitel 53xx from a remote/teleworker location, make sure port 69 is open on the corporate firewall to allow communication to the MiVoice Office 250 server.
- Mitel 52xx/53xx phones use DHCP option 128 for TFTP communication and DHCP option 129 for the call control address. Mitel 86xx phones use DHCP option 66 for TFTP communication and UDP discovery (broadcast) for the call control address (.cfg file).
- DB Programming provides the option to convert existing classic phone extensions to Mitel phone extensions.
- Preprinted paper designation cards are available for many models (such as the Mitel 5212 and 5224).

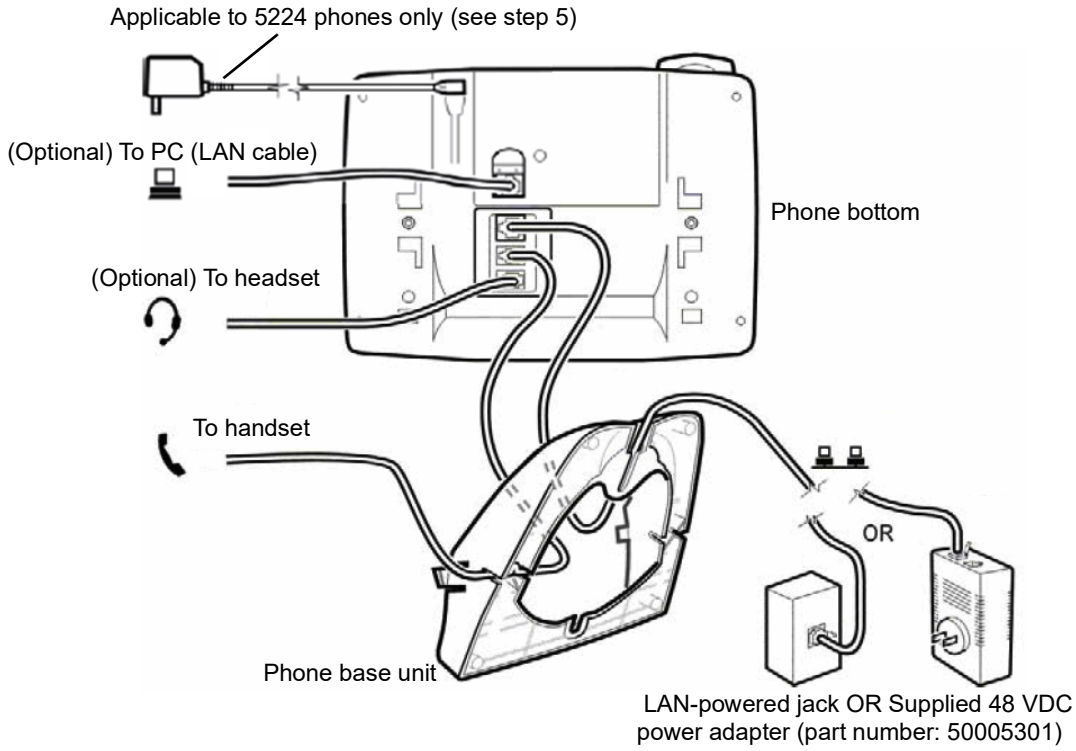
52xx/53xx Hardware Installation and Cabling Procedure

To install Mitel 52xx/53xx phones:

1. Plug the handset cord into the handset port and thread it through one of the slots on either side of the base unit. See Figure 3 on **page 259** for connection point examples.
2. Thread the LAN cable plug through the slot on the back of the installed base unit, and plug it into the LAN port on the back of the phone.
3. Plug the other end of the LAN cable into one of the following:
   - Phone port on Ethernet power adapter (power brick). Then, connect the Ethernet adapter to the LAN using a separate LAN cable. Plug the power adapter into an AC power outlet.
   - LAN-powered (PoE) jack. The LAN-powered phone can also provide power to any attached peripheral, such as a conference unit (reserved for future use).
4. (Optional) Plug the headset cord into the headset port on the back of the phone. Thread the headset cord through one of the slots on either side of the base unit, as shown for the handset cord in Figure 3 on **page 259**.

5. Optional for Mitel 5224 (reserved for future use): For the LAN-powered 5224 phone equipped with a peripheral (for example, the Mitel 5310 IP Conference Unit), plug the DC adapter cord (part number 51005172) into the power input and AC power outlet.
6. Do either of the following:
   - Continue to “Wall-Mounting Mitel 52xx/53xx Phones“ on **page 260**.
   - Continue to “Initial 52xx/53xx Automatic Phone Configuration (DHCP)“ on **page 261**.
   - Continue to “Initial 52xx/53xx Manual Phone Configuration“ on **page 263**.

Figure 3: Mitel 52xx/53xx IP Phone Connections



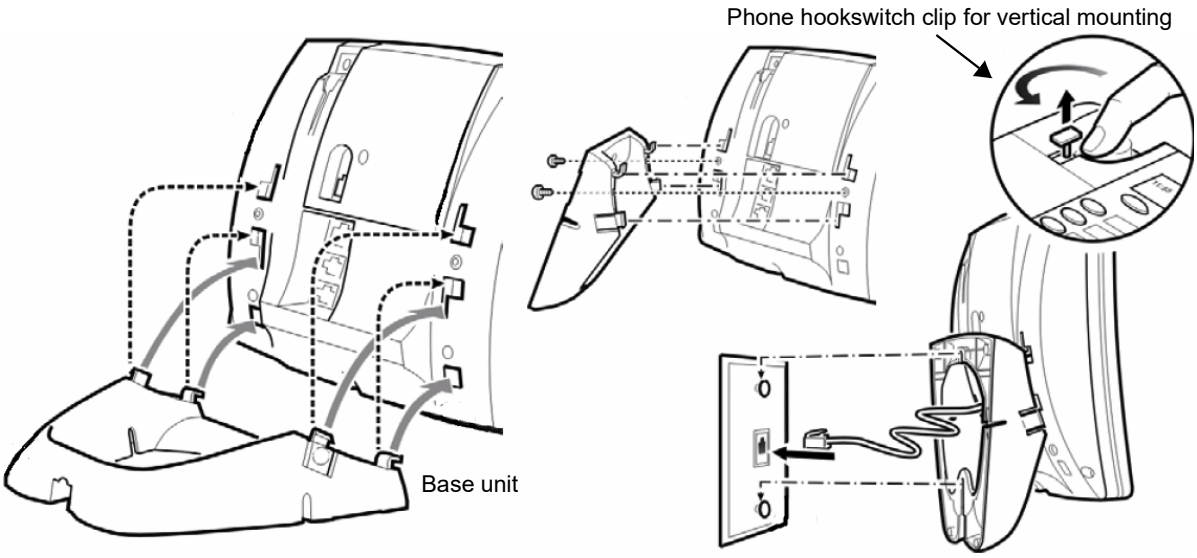
5224: 24VDC IP Power Adapter or 48VDC Ethernet/AC Power Adapter

5212: 48 VDC Power Adapter

Wall-Mounting Mitel 52xx/53xx Phones

The following procedure provides information on mounting the Mitel 52xx/53xx phones on a wall or on a wall plate. Use **Figure 4** as a guide.

Figure 4: Mitel 52xx/53xx Phone Wall-Mounting



To wall-mount the Mitel 52xx/53xx phones:

1. Select a location away from electrical cables, pipes or other items that could cause a hazard when inserting screws in the wall.

CAUTION

Equipment Damage Hazard: Make sure the wall is capable of supporting the weight of the phone. For additional security, mount the phone on a plywood backboard.

2. Turn the base unit so that the hooks on the flat edge of the base unit bend toward the top edge of the phone. Hinge the two upper hooks into the top set of notches and snap the two lower hooks into the middle set of notches on the back of the phone.
3. Secure the base unit using the two screws provided (found in a separate package with the phone).
4. Do one of the following:
   - *Mounting on a wall*:

     Insert two mounting screws 3 1/4 inches (8.25 cm) apart. Allow about 3/16 of an inch (0.5 cm) between the wall and screw heads for mounting the phone. Use #10 screws 1 to 1.5 inches (2.5 to 4 cm) in length for mounting screws and anchoring devices suitable for the wall.

   - *Mounting on a wall plate*:

   a. Plug the phone cord into the wall jack.

   b. Place the base unit on the wall plate post, and then push down until the unit is firmly seated.

5. Using your index finger, lift the clip (from the top) located below the hookswitch, turn the clip 180 degrees, and then let the clip snap into place to form a hook. The hook keeps the handset in the cradle when the phone is wall-mounted.

Mitel 52xx/53xx Phone Installation Troubleshooting

Table 44 lists general troubleshooting information for problems that could occur while installing Mitel 52xx/53xx phones.

Table 44: Troubleshooting Problems and Possible Solutions

Problem	Possible Solution
The UNLICENSED alarm appears on the phone display.	Verify that you have sufficient Category D licenses. To verify, dump the License Manager or use DB Programming.
The phone is not coming online.	Verify that the MAC Address is programmed correctly in DB Programming.
The phone is unable to load firmware.	Verify that the TFTP server is running on the 5000 (use the AWP to check status). Using OLM, verify that the TFTP server has the correct binary files. Refer to the diagnostics and troubleshooting information in the *MiVoice Office 250 Features and Programming Guide*.
The display name is partially cut off and/or a blinking cursor is displayed after the firmware is loaded.	There are two possible reasons for this type of error: • The system is down. • An incorrect MAC address is entered in DB Programming for the phone. A phone entered into DB Programming with an incorrect MAC address will still appear to download the initial configuration files; however, the display error message indicates that the phone is not configured in the system.
DHCP Discovery Option 128 Missing appears on the phone display.	Verify that the IP addresses entered in the phone for the Base server and TFTP server are correct (they must be the same IP address, if using the MiVoice Office 250 as the TFTP server).
The phone has been accidentally configured in “SIP Mode.” (SIP Mode is enabled by pressing *7 while the phone is powering on.) **Note:** In SIP Mode, a phone display may also show several scrolling Associated Press news clips	Power cycle the phone. While the phone is powering on, do one of the following: • Press and hold ***0**. • Press and hold ***78**. • Press and hold both Volume buttons. • Press and hold the button. After powering on the phone, configure as described in “Initial 52xx/53xx Manual Phone Configuration” on **page 1-263**.
ENHANCED SEC MODE appears on the phone display after the phone is powered on.	The phone must be returned for service. Contact your local provider or Mitel Technical Support for instructions.

Initial 52xx/53xx Automatic Phone Configuration (DHCP)

You can set up DHCP to provide additional information to avoid having to program each Mitel phone manually.

NOTE This procedure assumes the MiVoice Office 250 is up, configured, and running—this includes the desired number of Mitel 52xx/53xx phones configured in DB Programming.

To configure the Mitel 52xx/53xx phones automatically:

1. Make sure the MiVoice Office 250 is up and running, and then make sure you have done the following (if applicable):
   - Configured the Mitel 52xx/53xx phones in DB Programming (refer to the *MiVoice Office 250 Features and Programming Guide*, for details). Also, read “Hardware Revision Identification“ on **page 265**.

     If the phone is pointing to the MiVoice Office 250 and the phone has not been configured in DB Programming, the display appears to be downloading but nothing is actually happening. You must configure the phones in DB Programming before configuring the phone.
   - Check that the following DHCP options have been configured. Keep in mind that some DHCP servers, typically found in SOHO routers, do not support the ability for custom options. You can include all parameters within one option or you can configure several options.
     - *Setting a single option*: Using the Windows-provide **Option 43** or **Option 125**, type the following ASCII character string, after clicking once in the ASCII column:

       id:ipphone.mitel.com;sw_tftp=<*MiVoice Office 250 base server IP address*>;call_srv=<*MiVoice Office 250 base server IP address*>;vlan=<*VLANID*>;l2p=<*value*>;dscp=<*value*>

       Note that the entries in italics need to be replaced with those that are network-specific. For example, if the MiVoice Office 250 is using IP address 10.0.0.1; and the vLAN ID = 100; 12p = 6; and dscp = 46, then the typed ASCII character string would be as follows:

       id:ipphone.mitel.com;sw_tftp=10.0.0.1;call_srv=10.0.0.1;vlan=100;l2p=6;dscp=46
     - *Setting several options*:
       - **Options 128 and 129**: Option 128 provides the TFTP address, and Option 129 provides call control information. Both should be set to the IP address of the Base Server.
       - **Option 130**: Type the text string “MITEL IP PHONE” (case sensitive).
       - Mitel also has two additional Options to consider if putting these phones on a separate VLAN:
       - **Option 132**: Type the integer and set to “VLAN for IP Phone.
       - **Option 133**: Type the integer and set to “Priority on Vlan.” If using option 132, you must use option 133.
   - See the troubleshooting section on page 261 for any phone issues.
2. (Optional) You can connect a PC to the PC port on the back of the phone. If desired, thread the PC cable through the slot on the back of the installed base unit, as shown for the LAN cable in Figure 3 on **page 259**.
3. Continue to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*, for information on programming phones or other features in this release.

Initial 52xx/53xx Manual Phone Configuration

The following manual configuration procedure is required the first time only for each phone. All future updates to the phone are automatic. Provided the DB Programming option is set to enable the onboard TFTP server (refer to the *MiVoice Office 250 Features and Programming Guide*, you can provide future phone firmware updates by upgrading system software. The method to do this is through the Admin Session Web pages. To force a manual upgrade, you must reboot the phone using the volume up/down buttons, as described below.

NOTE This procedure assumes the MiVoice Office 250 is up, configured, and running—this includes the desired number of Mitel 52xx/53xx phones configured in DB Programming.

To configure the Mitel 52xx/53xx phones manually:

1. Make sure the MiVoice Office 250 is up and running, and then make sure you have done the following (if applicable):
   - Configured the Mitel 52xx/53xx phones in DB Programming (refer to the *MiVoice Office 250 Features and Programming Guide*. Also, read “Hardware Revision Identification“ on **page 265**.

     If the phone is pointing to the MiVoice Office 250 and the phone has not been configured in DB Programming, the display appears to be downloading but nothing is actually happening. You must configure the phones in DB Programming before configuring the phone.
   - See the troubleshooting section on page 261 for any phone issues.
2. While the phone is powering up, press both volume buttons and then release one of the volume buttons.
3. Using the dialpad buttons, enter **cfg (234)**.
4. Configure the network information for each phone. To backspace, press star (*); to enter a period, press (#).
   a. Press ***=YES** at the Network Parameters screen.
   b. Keep pressing **#=NO** until you get to the Modify Static Values screen.
   c. Press ***=YES** at the **Modify Static Values** screen. The Static Network Parms menu appears.
   d. Press the volume down button to scroll to **ICP IP Address**, and enter the MiVoice Office 250 Base Server IP address.
   e. Press the volume down button to scroll to **TFTP SVR IP Address**, and enter the MiVoice Office 250 Base Server IP address.
   f. Press the volume down button to scroll to **TFTP SVR Port**, and enter **69**. If left at 0, the default, the phone will first try port 20001 and then it will try port 69.
   g. Press the volume down button to scroll to **Store Changes**.
   h. Press ***=YES** to store the changes.
   i. Press ***=YES** at the Reboot Now display.

   When finished, the display shows the extension information and the date and time.

5. (Optional) You can connect a PC to the PC port on the back of the phone. If desired, thread the PC cable through the slot on the back of the installed base unit, as shown for the LAN cable in Figure 3 on **page 259**.
6. Continue to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*, for information on programming phones or other features in this release.

MItel 53xx Setup for Teleworking

This section details what must be done to configure the Mitel 53xx phones for teleworkers.

To configure the Mitel 53xx phones for teleworkers:

1. Make sure the MiVoice Office 250 is up and running, and then make sure you have done the following (if applicable):
   - Configured the Mitel 53xx phones in DB Programming (refer to the *MiVoice Office 250 Features and Programming Guide*. Also, read “Hardware Revision Identification“ on **page 265**. Make sure the phone is configured to use the NAT option.

     If the phone is pointing to the MiVoice Office 250 and the phone has not been configured in DB Programming, the display appears to be downloading but nothing is actually happening. You must configure the phones in DB Programming before configuring the phone.
   - Check that the main office firewall allows port 69 (TFTP) as well as the other call control ports mentioned in “Protocols and Ports” on page 125.
   - Check that the MiVoice Office 250 is configured with the NATed address for the Processor Module and the Processor Expansion Card.
   - Check that the main office firewall has the correct mapping for the appropriate ports. If needed, see “Network IP Topology“ on **page 417**.
   - Make sure all teleworker phones are not in a call configuration that uses P2P.
   - See the troubleshooting section on page 261 for any phone issues.
2. Hold down the volume keys and boot the phone to get in programming mode.
3. Using the menus, set the ICP and TFTP server address to be the NATed address of the MiVoice Office 250 Base Server.
4. Allow the phone to boot using DHCP, download the latest code and come online.
5. Power down the phone and deliver it to the user. The user simply plugs it in for it to boot up and begin working.
6. Continue to the *MiVoice Office 250 Features and Programming Guide*, for information on programming phones or other features in this release.

Installing Mitel 86xx IP Phones

IP phones are connected to a power source and to the network. There is no direct wiring from either the MDF or the MiVoice Office 250. All IP phones use a LAN connection. See the **Specifications** chapter for a list of supported phones.

This section covers the following primary topics:

- To begin installing IP phones, see page 265.
- For information on the physical interfaces for the IP phones, see page 268.
- For information on wall-mounting IP phones, see page 271.

For information on power supply compatibilities for the supported IP phones and devices, see the *IP Device Power Supply Compatibility Matrix*, part number 835.2688.

For programming information, refer to the *MiVoice Office 250 Features and Programming Guide*.

Installing IP Phones

For hardware revision information, see “Hardware Revision Identification” on **page 1-265**.

Hardware Revision Identification

Check the following areas to identify phone hardware types:

- **Phone Web Interface:** The hardware type is displayed in the Firmware Information field under Status – Firmware – Hardware Type.
- **LCD (Liquid Crystal Display):** The hardware type is displayed on the bottom line of the LCD on startup.
- **Product ID and Revision:** The part number and revision are displayed on the labels on the bottom of the phone.
- **Message Lamp LED (Light-Emitting Diode) Pattern:** The hardware type is identified by the message lamp LEDs during startup. The user must reset the phone first.
- **Ethernet Ports:** The number of Ethernet interfaces can be used to identify the hardware type.

Table 45, “Hardware Revision Identification,” on **page 265** lists the hardware types and identification methods.

Table 45: Hardware Revision Identification

Phone	Hardware Type	LCD	Product ID (p/n)	LED Pattern	Ethernet Port
8600 — original	8600	Not Available	550.8600	The phone LEDs flash in a left-right alternative blinking pattern.	1 Port
8620 — original	8620	8620	550.8620	Not Available	1 Ports
8622 — original	8622	8622	550.8622	Not Available	2 Ports
8662 — original	8662	8662	550.8662	Not Available	2 Ports

Table 45: Hardware Revision Identification (Continued)

Phone	Hardware Type	LCD	Product ID (p/n)	LED Pattern	Ethernet Port
8662 — enhanced	8662	8662	550.8662E	Not Available	2 Ports
8600 — enhanced, redesigned	8600v2	Not Available	550.8600	The phone performs a blinking LED pattern of the XFER, CNF, MSG, FWD, and DND button LEDs.	1 Port
8622 – enhanced, redesigned	8622v2	8622v2	550.8622	Not Available	2 Ports
8662 – enhanced, redesigned	8662v2	8662v2	550.8662E	Not Available	2 Ports
8622 — PoE Class 2	8622	8622	550.8622P	Not Available	2 Ports
8662 — PoE Class 2	8662	8662	550.8662P	Not Available	2 Ports

To install 8600-series IP phones:

See the following illustrations for connection point examples:

- Mitel 8660 on **page 268**
- Mitel 8600, 8620/8622, and 8662 on **page 269**
- Mitel 8690 on **page 270**

1. Make sure the MiVoice Office 250 is up and running. Also, read “Hardware Revision Identification” on page 265.
2. Connect one end of the Ethernet cable to a PC port on the back of the phone. Connect the other end of the cable to the PC, if applicable.

   For the Mitel 8660, use unbooted Ethernet cables. The Mitel 8660 does not support booted cables because the plastic “protective” ends on booted Ethernet cables get in the way when routing the cable away from the phone.
3. Connect the power supply, using one of the following methods. Use CAT5 or better cable for all LAN jack/plug connections in this procedure.

   Depending on the IP devices to be used, there are several power supplies available. For a list of power supply compatibilities for the supported IP phones and devices, see the *IP Device Power Supply Compatibility Matrix*, part number 835.2688.

   - *USA/European/Universal Power Supply using a LAN Jack*:

   a. Connect the LAN/POWER jack on the baseplate to the KS/SLA jack on the 24 VDC adapter.

   b. Using another cable, connect the LAN jack on the 24 VDC adapter to the LAN.

   c. Connect the 24 VDC adapter to a 110 V wall socket. Mitel currently offers the following power supply kits.

Table 46: Power Supply Kits

Part Number	Contains
828.1642	• U.S. power supply, part number 806.1113 • Network cable, part number 813.1790
828.1660	• Universal power supply, part number 806.1119* • Network cable part number 813.1790
* Requires a standard IEC (International Electrical Cord) power connector.	

- *Power over Ethernet (PoE)* hub *(LAN Jack)*: The Red Hawk PowerSense™ modular, in-line PoE hubs (for multi-port) are available through the Mitel DataNet CommSource division. The power hub provides power to IP devices without the use of an external power supply for each device. For details, see the *Red Hawk PowerSense Power over Ethernet Installation Manual*, part number 935.0400.

a. Pug one end into the LAN/POWER (RJ-45) jack on the IP device.

b. Plug the other end of the cable into the POWER/DATA (RJ-45) jack on the PoE connector.

- *Universal Power Supply using a Barrel Jack:*

a. Using CAT5 or better cable, connect the LAN/POWER jack on the baseplate directly into the network. Do not connect this cable to the PC 1, PC 2, or PC 3 jack.

b. Connect the adapter to an AC voltage power source and connect the barrel plug on the power adapter to the barrel jack on the baseplate (if available). The center of the barrel jack is positive. The minimum power requirements for the IP device is 24VDC at 500mA (using a Universal power supply) and 24VDC at 400mA (using a customer-provided power supply).

4. While the phone is connecting to the IP Resource application, the display shows the following connection-state information:

- Ethernet address
- IP address
- Subnet mask
- Gateway address
- Address of the IP resource to which it is connecting.
- Station identification (extension number and assigned username) for several seconds.
- The LCD changes to the appropriate display, depending on its current status. See the tables beginning on **page 276** for an explanation of status displays.

NOTE

If the phone appears to be stuck in “Waiting For Download” mode, try to ping the phone IP address. If you cannot ping the phone, assign a valid BOOTP IP address from a BOOTP server to the device Ethernet address. Also, change the default IP address, subnet mask, and default gateway so that the device receives a better default when it reconnects to the IPR application after the download. When waiting for a download, the IP device will attempt BOOTP, even if BOOTP is disabled in IPR application programming. Power cycle the IP device, then download the software.

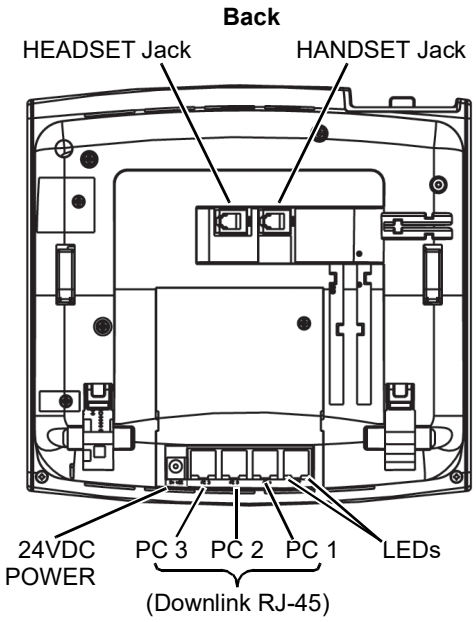
The IP Resource application does not provide a BOOTP server and requires a static IP address or the use of either a third-party BOOTP server or a DHCP server for IP address assignment.

Physical Interfaces

See the sections that follow for information on the supported phone physical interfaces to the system. See page 265 for the installation procedure. See page 271 for information about wall-mounting the phones.

Mitel 8660

The Mitel 8660 is supported by both 10Base-T and 100Base-TX interfaces, but not 100Base-T4, which uses four pairs of wires. See the illustration at right.



Mitel 8600

The Mitel 8660 includes the following connections:

- **LAN/POWER Jack**: Connects to a network hub or a switch and gets power from pins 7 and 8.
- **POWER Jack**: Connects to a 24 VDC external power supply. Use only one power source to connect the power supply.
- **PC 1-3 Ports**: Connect to PCs or any other Ethernet devices. The speeds on single uplink and three downlink ports are independent from each other and auto-negotiable.
- **Handset Jack**: Connects to a handset.
- **Headset Jack**: Connects to a headset.
- **LEDs**: There are two indications on each port:
  - *Link Status*: When the link is valid, the green LED on the left side of the port is lit. When there is receive or transmit activity on the link, this LED flashes.
  - *10/100 Mbps*: When the link is connected at a speed of 100 Mbps, the green LED on the right side of the port is lit. The MiVoice Office 250 operates ***only*** on 100Base-T LANs.

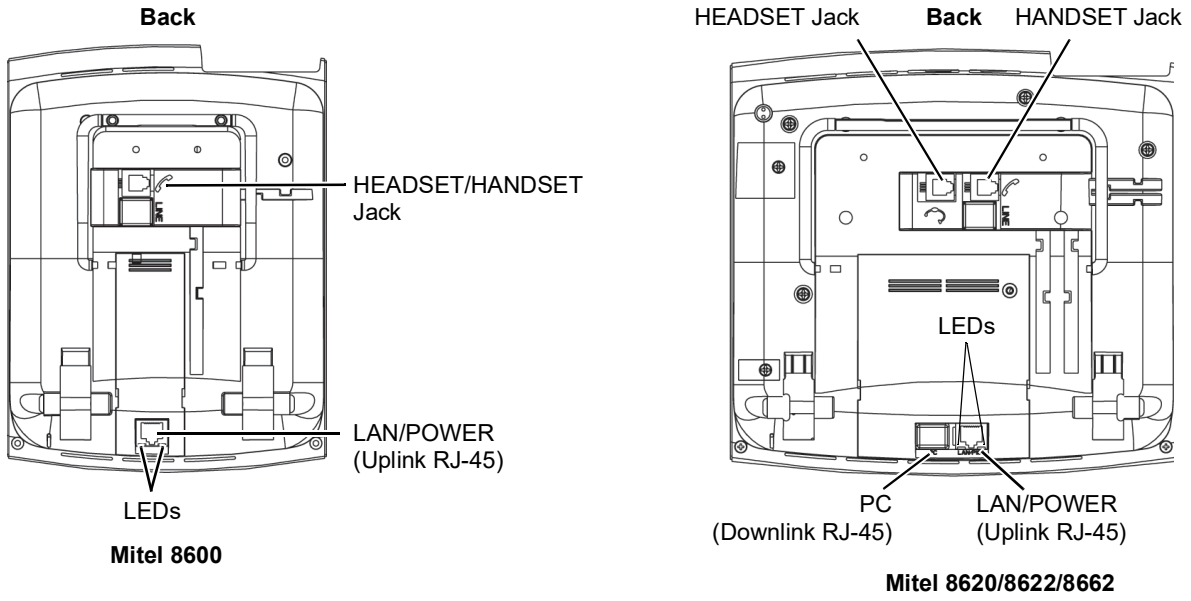
For information about programming a Mitel 8660, refer to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*.

Mitel 8600, 8620/8622, and 8662

Mitel 8600, 8620/8622, and 8662 multi-protocol phones operate the same as Mitel 8500, 8520, and 8560 phones, respectively. These phones support IP Mode or SIP mode using 100Base-TX interfaces. All phones are configured to use IP by default. For details, see page 275.

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform. See page 275 for changing from SIP mode to IP mode.

Figure 3: Connection Interfaces for 8600 Series Multi-Protocol Phones



The Mitel 8600 series phones include the following connections:

- **Headset/Handset Jack**: Connects to a headset or handset.
- **LAN/POWER Jack**: Connects to an external power supply or a network hub/switch.
- **PC Port**: (Mitel 8622 and 8662 only) Connects to a PC or any other 100Base-T Ethernet device. The speeds on single uplink and downlink port are independent from each other and auto-negotiable.
- **LEDs**: There are two indications on each port:
  - *Link Status*: When the link is valid, the green LED on the left side of the port is lit. When there is receive or transmit activity on the link, this LED flashes.
  - *10/100 Mbps*: When the link is connected at a speed of 100 Mbps, the green LED on the right side of the port is lit. The MiVoice Office 250 operates *only* on 100Base-T LANs.
- **DRAM**: Dynamic Random Access Memory located inside the phones. Mitel 8600, 8620, and 8662 support 8 MB, and Mitel 8622 supports 16 MB (only one slot is occupied). Although two DRAM slots exist, you cannot upgrade the memory by adding an extra memory stick. The second slot is not used.

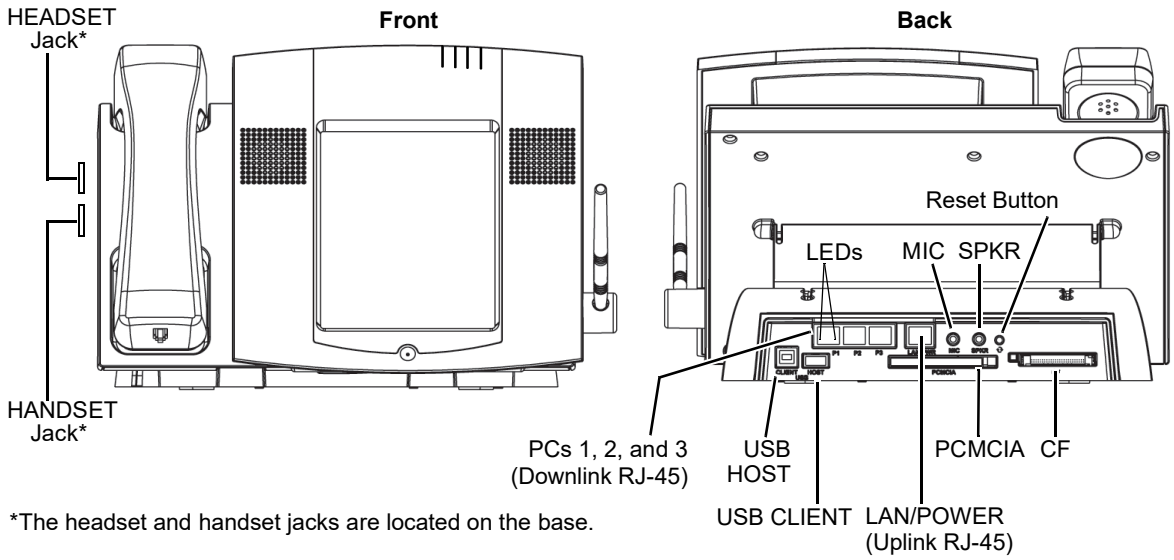
For information about programming Mitel 8600 series phones, refer to the *MiVoice Office 250 Features and Programming Guide*.

Mitel 8690

The Mitel 8690 multi-protocol phone has a touch screen that displays feature, dialpad, and menu buttons. It supports IP or SIP mode using 100Base-TX interfaces. The Mitel 8690 is configured to use IP mode by default. For details, see page 275.

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform. See **page 275** for changing from SIP mode to IP mode.

Figure 4: Mitel 8690 Connection Interfaces



*The headset and handset jacks are located on the base.

The Mitel 8690 includes the following connections points and options:

- **LAN/POWER Jack**: Connects to an external power supply or a network hub/switch.
- **PC Ports 1-3**: Connect to PCs or any other 100Base-T Ethernet devices. The speeds on the single uplink and three downlink ports are independent from each other and auto-negotiable.
- **Headset/Handset Jack**: Connects to a headset or handset.
- **SPKR Jack**: Connects to a speaker and supports mono for IP phone and stereo for Windows CE playback.
- **MIC Jack**: Connects to a microphone equipped with a 3.5mm plug.
- **Reset Button**: Resets the phone when pressed and released quickly.

NOTE If you press and hold the Reset button until the four LEDs on the front of the phone (Message Lamp) start blinking, the system defaults the phone configuration (such as the configuration TFTP URL).

- **CF Port**: Connects to a matchbook-sized compact flash-type memory card that can be used for copying music, pictures, etc. to the Mitel 8690. The card also can be used for additional memory.

- **PCMCIA Port**: Connects to a credit card-sized memory card manufactured to the standards of the Personal Computer Memory Card International Association.
- **USB HOST Port**: Connects to a standard Universal Serial Bus keyboard, mouse, etc. For details about how to connect a keyboard and/or mouse, see the applicable *Model 8690 Phone User Guide (IP Mode or SIP mode)*, part number 550.8116 or 550.8025, respectively.
- **USB CLIENT Port**: Connects to a host computer that runs Microsoft ActiveSync®.
- **LEDs**: There are two indications on each port:
  - *Link Status*: When the link is valid, the green LED on the left side of the port is lit. When there is receive or transmit activity on the link, this LED flashes.
  - *10/100 Mbps*: When the link is connected at a speed of 100 Mbps, the green LED on the right side of the port is lit. The MiVoice Office 250 operates *only* on 100Base-T LANs.
- **DRAM**: Located inside the phone. Supports 128 MB (only one slot is occupied). Although there are two DRAM slots, you cannot upgrade the memory by adding an extra memory stick. Currently, the second slot is not used.

For information about how to program a Mitel 8690, refer to the *MiVoice Office 250 Features and Programming Guide*.

To use the Mitel 8690, you must first install the 8690 client application. refer to the *MiVoice Office 250 Features and Programming Guide*, for details.

Wall-Mounting 86xx IP and IP PhonePlus Phones

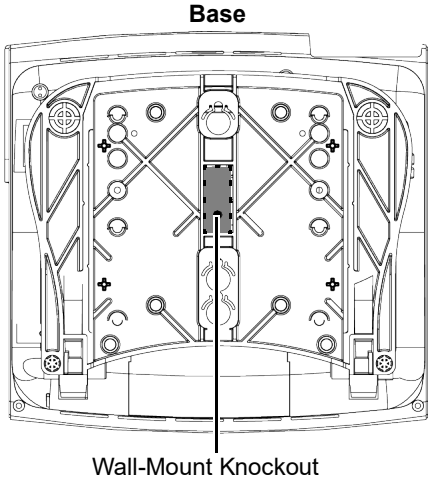
See the following sections for information about wall-mounting your specific phone type.

Mitel 8600 Series Phones

> **NOTE** Wall-mounting the Mitel 8600 series phones is ***not*** recommended if the user will be connecting and disconnecting cords (handset, headset, line) and/or cables (Ethernet, USB, power supply) frequently. To connect and disconnect these cords and cables from a wall-mounted phone, users will need to remove the phone from the wall.

Because there are two types of bases used on the Mitel 8600 series phones, two sets of instructions for wall-mounting the phones are included here. Determine which type of base the phone has and follow the appropriate instructions.

For best results when wall-mounting the Mitel 8600 series phones, Mitel strongly recommends that you use the two plastic rivets shipped with the phone. Once installed, the plastic rivets lock the components into place, providing additional stability for wall-mounted phones.

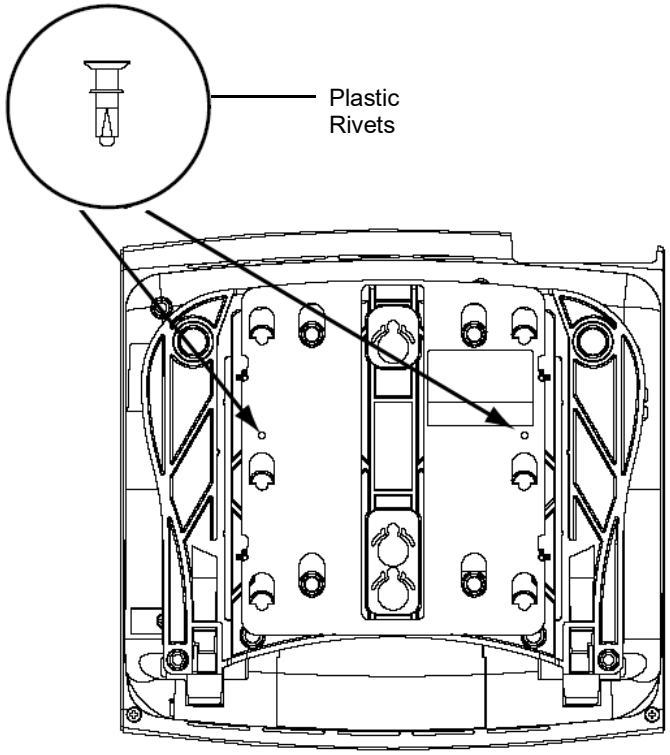


Mitel 8600 series phones are shipped with a plastic adhesive label that covers the rectangular wall-mount knockout on the base of the phone. A portion of the adhesive label must be removed to use the wall-mount knockout.

To wall-mount the Mitel 8600 series phones, you need an industry-standard wall-mount bracket (sometimes called a wall jack assembly) and a four-inch line cord. Both of these items are available through the Mitel DataNet CommSource division.

To prepare the phone for wall-mounting (regardless of base type):

1. Remove the metal plate from the base of the phone by removing the two plastic rivets to unlock it from the base.

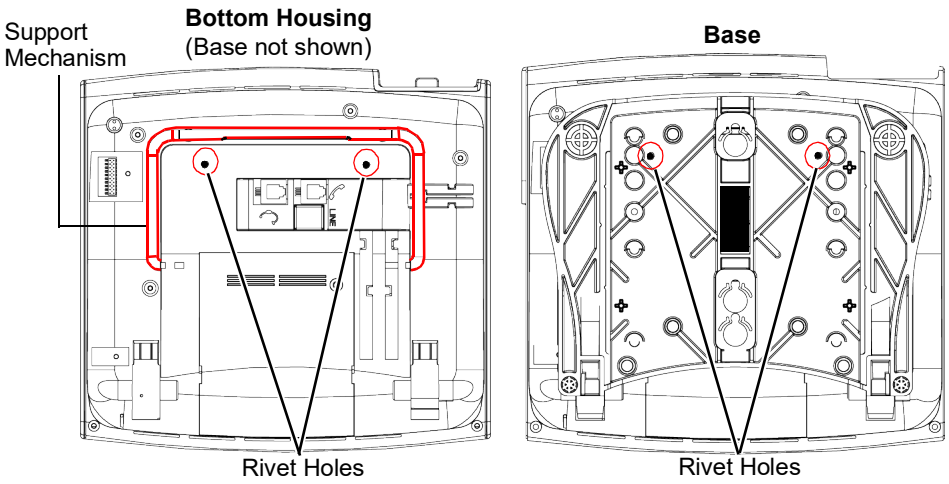


2. Slide the cover off. The two plastic rivets are not included on phones built prior to May 2005.
3. Use a sharp blade to cut out the rectangular portion of the adhesive label covering the wall-mount knockout (see diagram at right).

The following instructions apply to phones that have a support mechanism—see the following illustration—that fits into holes in the base.

To wall-mount the Mitel 8600 series phones that have a support mechanism:

1. Collapse the support mechanism completely and secure it against the bottom housing of the phone, as shown in the following illustration.

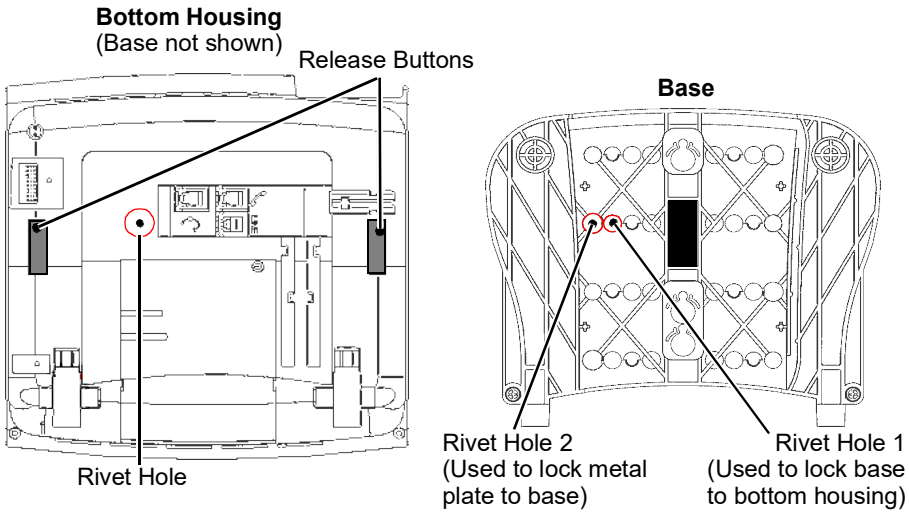


2. Close the base completely so that it lies flat against the phone. (The two rivet holes on the base align with the two rivet holes on the bottom housing of the phone.)
3. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
4. Insert the two plastic rivets through the base and press them firmly into the holes on the bottom housing of the phone. This locks the base to the phone.
5. Test the base to verify that it is securely locked to the phone.
6. Re-attach the metal plate to the phone base.
7. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.
8. Flip the handset hanger down to the horizontal position and lock it into place. This will hold the handset in place.

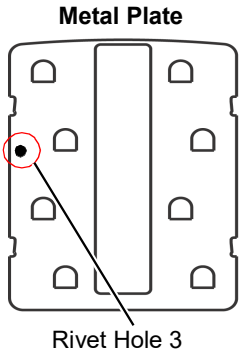
The following instructions apply to phones that have release buttons located on the bottom housing of the phone. To disengage the locking mechanism in the base, push in both buttons. To lock the base in position, release the buttons.

To wall-mount Mitel 8600 series phones with release buttons on the bottom housing:

1. Close the base completely so that it lies flat against the phone. (Rivet hole 1 on the base aligns with the rivet hole on the bottom housing of the phone.)



2. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
3. Insert one of the plastic rivets in rivet hole 1 on the base (see the illustration above) and press it into the rivet hole on the phone. This locks the base to the phone.
4. Test the base to verify that it is securely locked to the phone.
5. Reattach the metal plate to the phone base.
6. Insert the remaining plastic rivet through rivet hole 3 (as shown at right) in the metal plate and press it firmly into rivet hole 2 in the base. This locks the metal plate to the base.
7. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.
8. Flip the handset hanger down to the horizontal position and lock it into place. This holds the handset in place.



Changing Protocol Modes: IP and SIP

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

The Mitel 8600, 8620/8622, 8662, and 8690 multi-protocol phones support Internet Protocol (IP) or Session Initiation Protocol (SIP) mode. Details about each of the modes are described in the **Specifications** chapter. For complete information about supported features, buttons, and light-emitting diode (LED) indications for multi-protocol phones in IP or SIP mode, see the *Multi-Protocol Phones: Supported Features, Buttons, and LED Indications* document, part number 835.2840.

To change the modes on Mitel 8600, 8620/8622, and 8662:

- **Dialpad**: *To switch to IP mode*, hold down 5 and 6 on the dialpad while powering up.

IMPORTANT	The preferred method for changing between SIP and IP mode is through the Administrative Web Portal. For details discussed in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*.

- **Web Interface**: Change the operation mode through the Web interface to the phone.
- **TFTP Server**: Change the configuration files using the TFTP server.

To change the modes on the Mitel 8690:

Change the operation mode through the Web interface.

Reading 86xx IP Phone Status Displays

While the boot code is loading, you will see the displays shown in the following table. The top lines on the display change, as shown in the chart, and the bottom lines always contain the IP address and the Ethernet address.

Several displays have a "/" character at the end that is a rotating bar to indicate that the device is working.

Firmware Download Progress Displays

Several changes have been made to the firmware image update process. These changes only apply when the phone is running version 1.2.7 or later firmware (not when upgrading from 1.1.5 to 1.2.7, for example).

- Two copies of the firmware image are kept in case one gets corrupted during a failed download attempt (this does not apply to the Mitel 8600 phones). This may cause the firmware update process to take more time than before.
- Additional messages are displayed on the LCD during the firmware download process to let the user know the status of the firmware download and to discourage the unintentional interruption of the process—for example, unplugging the phone because nothing appears to be happening. **Table 47** shows the update progress displays.

Table 47: Firmware Download Progress Displays

Display	Description
DOWNLOAD IN PROGRESS	The phone is downloading the image from the TFTP server.
ERASING PRIMARY FLASH	The phone is erasing primary flash. The phone is preparing to program the primary flash with the new software image. This message is usually only shown briefly.
ERASING BACKUP FLASH	The phone is erasing backup flash. The phone uses the backup flash as a place holder when downloading the image or as a backup storage to restore the image in case the primary flash is corrupted. This message is usually shown briefly.
WRITING IMAGE TO PRIMARY FLASH	The phone is programming the image to the primary flash.
WRITING IMAGE TO BACKUP FLASH	The phone is programming the image to the backup flash.

Table 47: Firmware Download Progress Displays (Continued)

Display	Description
FIRMWARE DOWNLOAD SUCCESS	The phone has successfully updated the firmware. The phone will reset.

If an error occurs during the firmware download process, the phone displays an error message as described in “Download Error Message Displays” on **page 1-280**.

IP PhonePlus and Mitel 8660

Table 48: IP PhonePlus and Mitel 8660 Progress Displays

Display	Status
WAITING FOR DOWNLOAD	**Awaiting Download:** The boot code is waiting for a code download.
ERASING FLASH	**Programming State - Erasing Flash Memory:** The boot code is receiving object code and programming the flash memory. The phone first erases the flash memory and then displays this message.
PROGRAMMING CODE	**Programming State - Programming Flash Memory:** After erasing the flash, the phone programs the flash memory.
DOWNLOAD COMPLETED	**Completed State:** The download completed successfully.
HARDWARE FAILURE FLASH ERASURE	**Error State – Flash Erasure:** The boot code encountered an error in erasing the flash chips. If this occurs, try two or three downloads before returning the phone for repair.
HARDWARE FAILURE FLASH PROGRAM	**Error State – Flash Program:** The boot code encountered an error while programming the flash chips. If this occurs, try two or three downloads before returning the phone for repair.
HARDWARE FAILURE MEMORY MAP	**Error State – Memory Map:** The hardware memory-map functions are not working properly. After two or three failed attempts, you should send in the phone for repair.
DOWNLOAD ABORTED FILE TOO BIG	**Error State – File Too Big:** This display appears if you attempt to download a file that is too large for the flash memory. This error condition might indicate an incorrect or corrupted file.
DOWNLOAD ABORTED FILE CORRUPTED	**Error State – File Corrupted:** This display appears if you attempt to download a file that is corrupted or that gets corrupted during the transmission. This error condition might occur during poor network conditions.

Mitel 8620/8622, 8662, and 8690

The following information does not apply to a non-display phone, such as the Mitel 8600.

Table 49: Mitel 8620/8622, 8662, and 8690 Progress Displays

Display	Status
DOWNLOAD IN PROGRESS	**Download State:** The file is being downloaded to the phone.
FIRMWARE DOWNLOAD SUCCESS	**Completed State:** The download completed successfully. The phone will reboot momentarily.
ERROR TFTP FAIL	**Error State – TFTP Fail:** The phone failed to contact the TFTP server, or the file requested is not available at the server. It could also indicate that the TFTP server does not have permission for the phone to retrieve the file.
ERROR TFTP FAIL START	**Error State – TFTP Fail Start:** The TFTP server failed to start due to lack of resources. If this occurs, reboot the phone.

Table 49: Mitel 8620/8622, 8662, and 8690 Progress Displays (Continued)

Display	Status
ERROR BAD FLASH	**Error State – Bad Flash:** The phone is unable to program the new firmware to the flash device. If this occurs continuously, return the phone for repair.
ERROR TFTP TIMEOUT	**Error State – TFTP Timeout:** The TFTP protocol timed out due to network problems, such as bad network traffic.
ERROR WRONG VERSION	**Error State - Wrong Version:** The firmware version of the file being downloaded does not match the version in the version field in the configuration file (xxxx_image_ver or 8690_dsp_image_ver).
ERROR WRONG PLATFORM	**Error State – Wrong Platform:** The file requested does not match the phone type.
ERROR MISC ERROR	**Error State – Miscellaneous Error:** A miscellaneous error, such as file corruption, occurred.
ERROR NOT GZIP	**Error State – Not Gzip:** The file requested is not in the correct format.
ERROR INVALID FILE ID	**Error State – Invalid File ID:** The file does not contain the necessary version information and platform type.

While the application code is executing, you will see the following display as it moves through several states. After the phone is connected to the IP card, the default display appears on the LCD.

Several displays have a "/" character at the end that is a rotating bar to indicate that the device is working.

IP PhonePlus and Mitel 8660/8620/8622/8662/8690

The following information does not apply to a non-display phone, such as the Mitel 8600. Also, because the Mitel 8620/8622 has a two-line LCD, this phone displays only the first two status lines.

Table 50: IP PhonePlus and Mitel 8660/8620/8622/8690 Progress Displays

Display	Status
INITIALIZING TRYING BOOTP... NO RESPONSE TRYING DHCP... SUCCESSFUL /	**Initializing:** The application code is initializing the IP stack. For the majority of the initialization time, the IP phone is waiting for a BOOTP or DHCP response. For each enabled option (BOOTP, DHCP, or both), the phone screen displays a message indicating the type of response it is attempting to receive. • If the IP phone receives a BOOTP or DHCP response, it uses the IP address, subnet mask, and gateway address from the response. • If the IP phone times out, it uses its default IP address, subnet mask, and gateway address from its internal database. • If at least one of these options is enabled and the phone is not able to receive an IP address, the phone displays NO RESPONSE for five seconds. • If the phone tries BOOTP and fails, but then tries DHCP and succeeds, the display does not pause for five seconds.
(Sheet 1 of 3)	

Table 50: IP PhonePlus and Mitel 8660/8620/8622/8690 Progress Displays (Continued)

Display	Status
ID: XXXXXXXXXXX MY IP ADDRESS: XXX.XXX.XXX.XXX FINDING LOCAL IP CARD ON UDP PORT XXXXX /	**Finding IP Card (Module) Address:** The application code is attempting to find a local IP resource (MiVoice Office 250) by broadcasting on the LAN. The first line contains the hexadecimal Ethernet address of the phone, while the second and third lines display the phone's IP address. The remainder of the display phone screen indicates that the phone is attempting to find an IP address.
ID: XXXXXXXXXXX MY IP ADDRESS: XXX.XXX.XXX.XXX FINDING IP CARD: XXX.XXX.XXX.XXX UDP PORT XXXXX /	**Finding IP Card (Module):** The application code is attempting to find a local IP resource (MiVoice Office 250) by using the reply from the LAN broadcast. The first line contains the hexadecimal Ethernet address of the phone, while the second and third lines inform the user of the phone IP address. The rest of the LCD panel informs the user that the phone is attempting to find the IP address and its port number.
ID: XXXXXXXXXXX MY IP ADDRESS: XXX.XXX.XXX.XXX CONNECTING TO: XXX.XXX.XXX.XXX TCP PORT XXXXX /	**Connecting to IP Card:** The application code is attempting to connect to an IP resource (MiVoice Office 250). The first line contains the hexadecimal Ethernet address of the phone. The second line contains the IP phone current IP address. The fourth line contains the IP address to which the IP phone is attempting to connect. The sixth line indicates the number of unsuccessful connection attempts.
ID:XXXXXXXXXXX XXX.XXX.XXX.XXX CONNECTED TO: XXX.XXX.XXX.XXX WAITING FOR CALL PROCESSING/	**Connected to IP Card:** The application code has made an IP connection to an IP resource (MiVoice Office 250). The IP phone maintains this display until it receives a confirmation from call processing. Some possible problems are that call processing is not running, the module or circuit is not programmed in the call processing database correctly, or the phone is connected to the wrong type of system. The first line contains the hexadecimal Ethernet address of the phone. The second line contains the IP phone current IP address. The fourth line contains the IP address to which the IP phone is connected.
ID: XXXXXXXXXXX LOCAL IP CARD NOT FOUND REMOTE IP CARD NOT PROGRAMMED /	**IP-Card-Not-Found Error Message:** After an attempt to find the local IP resource (MiVoice Office 250) through broadcast messages, the phone would normally attempt to contact the Remote IP resource (MiVoice Office 250) IP address. However, the phone does not have this field programmed; you can program this field through the Self-Programming Mode or the Web/Telnet interfaces. The display will show this error message for five seconds.
ID: XXXXXXXXXXX COULD NOT CONTACT IP CARD XXX.XXX.XXX.XXX USING UDP PORT XXXXX /	**UDP-Failed Error Message:** If the phone did not receive a response via UDP after attempting to contact the IP resource (MiVoice Office 250), the display will show this error message for five seconds.
ID: XXXXXXXXXXX COUNT NOT CONTACT IP CARD XXX.XXX.XXX.XXX USING TCP PORT XXXXX /	**TCP-Failed Error Message:** If the phone did not receive a response via TCP after attempting to connect to the IP card, the display will show this error message for five seconds.

(Sheet 2 of 3)

Table 50: IP PhonePlus and Mitel 8660/8620/8622/8690 Progress Displays (Continued)

Display	Status
ID: XXXXXXXXXXXX XXX.XXX.XXX.XXX DUPLICATE IP ADDRESS AT MAC ADDRESS: XXXXXXXXXXXX	**Duplicate-IP Error Message:** If the phone determined that some other device is using the IP address that it is trying to use, the display will show the error message for thirty seconds. The display shows the IP address the phone is trying to use, that the address is in use by another device, and the MAC address of that device.
DSP FAILURE	**DSP Failure:** The application code displays the DSP Failure message if there was an error (bus error) while accessing the DSP. If this occurs, try two or three downloads before returning the phone for repair. After two or three failed attempts, send in the phone for repair.
DSP NOT RUNNING DSP NOT RUNNING	**DSP Not Running:** The application code displays the DSP Not Running message if the phone was able to access the DSP but has not received any response from the DSP (no online interrupt). If this occurs, try two or three downloads before returning the phone for repair. After two or three failed attempts, send in the phone for repair.
WAITING FOR ETHERNET ADDRESS	**Waiting for Ethernet Address:** The application code has detected that the phone does not have a valid Ethernet address. If this occurs, try two or three downloads before returning the phone for repair. After two or three failed attempts, send in the phone for repair.
	(Sheet 3 of 3)

Download Error Message Displays

The phone displays error information concerning the status of the last TFTP application image update (firmware or boot code) not only on the LCD but also on the Web interface. The error message is displayed under Status\Image Upgrade\Last Download Error. **Table 51** lists the possible error messages.

Table 51: Image Download Error Message Displays

Display	Description
ERROR TFTP FAIL	The image download process failed because of one of the following problems: • The phone cannot contact the TFTP server. • The TFTP server does not have the software image file. • The TFTP image URL contains the wrong path. • The TFTP server does not have the permission to read the software image file.
ERROR TFTP FAIL START	The phone failed to start TFTP client. This problem could be due to an insufficient file descriptor for TFTP, etc.
ERROR BAD FLASH	The image download process failed because of a faulty flash memory device or the flash file system is corrupted. If the user encounters this error, replace the device.
ERROR TFTP TIMEOUT	The phone failed to communicate with the TFTP server. This error could be due to bad network connectivity.
ERROR WRONG VERSION	The image contains an unexpected software version.
ERROR WRONG PLATFORM	The image contains an unexpected platform identifier.

Table 51: Image Download Error Message Displays (Continued)

Display	Description
ERROR MISC ERROR	Miscellaneous errors include a corrupted file system, unrecognized error by the TFTP library, etc.
ERROR NOT GZIP	The image is not in GZIP format.
ERROR INVALID FILE ID	The image does not contain the appropriate file identifier.
ERROR INCOMPATIBLE HW	The image contains the wrong hardware ID.
ERROR FILE TOO BIG	The file that the phone downloaded for upgrade is too large.
ERROR BOOT IMAGE CORRUPT	The boot image downloaded is corrupt.
ERROR ALLOCATE FAILURE	The phone was unable to allocate enough memory to download the image.
ERROR FAILED VERIFY	The image written to flash does not match the file downloaded. A write error occurred.

Where to Program IP Resources

Program IP resources in DB Programming. Refer to the *MiVoice Office 250 Features and Programming Guide*.

86xx Firmware

This section provides information on the multi-protocol phone firmware, including uploading firmware and phone firmware compatibilities.

Phone Firmware Compatibilities

Table 52 lists the compatibilities of the multi-protocol phone firmware.

Table 52: Multi-Protocol Phone Firmware Compatibility Matrix

Phone (Hardware Type[a])	Description	Firmware ✓=supported			
		1.01S	1.0.31/ 1.0.33	1.1.5	2.0.03 or Later
Mitel 8600 (8600)	8 MB RAM, 2 MB flash memory, original	✓	✓	✓	✓
Mitel 8620 (8620)	8 MB RAM, 4 MB flash memory, original	✓	✓	✓	✓
Mitel 8622 (8622)	16 MB RAM, 4 MB flash memory, original	✓	✓	✓	✓
Mitel 8662 (8662)	8 MB RAM, 4 MB flash memory, original	✓	✓	✓	✓
Mitel 8662 (8662)	16 MB RAM, 4 MB flash memory, enhanced	✓	✓	✓	✓
Mitel 8622 (8622v2)[b]	16 MB RAM, 4 MB flash memory, enhanced, redesigned				✓
Mitel 8662 (8662v2)[b]	16 MB RAM, 4 MB flash memory, enhanced, redesigned				✓

a. See page 265 for details about hardware types.
b. Cannot be downgraded to v1.1.5 or earlier. These phones do ***not*** function with earlier firmware versions.

Firmware Update Changes

When upgrading an 8620, 8622, or 8662 phone, use the following version strings. For instructions on how to update firmware, see “Updating Firmware“below.

- From 1.01S to 1.0.31

  86xx_image_ver: 1.0.31 (where the 86xx indicates the model type, such as 8622)

  86xx_image_url: 192.168.200.202/86xx/86xx_1_0_31.bin

- From 1.01S or 1.0.31 to 1.0.33

  86xx_image_ver: 1.0.33

  86xx_image_url: 192.168.200.202/86xx/86xx_1_0_33.bin

- From 1.0.33 and earlier to 1.1.5

  86xx_image_ver: HWID 1 1.1.5

  86xx_image_url: 192.168.200.202/86xx/86xx_1_1_5.bin

  If upgrading an 8622 from 1.01S to 1.1.5, use the following image version string: 8620_image_ver: 8620 HWID 1 1.1.5.

- From 1.1.5 and earlier to 2.0.06

  86xx_image_ver: HWID 2 2.0.06

  86xx_image_url: 192.168.200.202/86xx/86xx_2_0_06.bin

- For future releases (2.0 and later)

  itphone_image_ver: v.v.vv (where v.v.vv indicates the firmware version, such as 2.0.06)

  itphone_image_url: 192.168.200.202/itphone/itphone_v_v_vv.bin (where v_v_vv.bin indicates the firmware version, such as 2_0_06)

  After a phone is updated to v2.0.06 or later firmware, the 86xx_image_ver and _url strings are no longer necessary. There are only two lines in the configuration files that need to be modified (itphone_image_ver and itphone_image_url), and one image to be put on the TFTP server to update all of the multi-protocol phones (excluding the Mitel 8690).

Updating Firmware

Always upgrade all multi-protocol IP phones to the latest firmware release. This will minimize potential conflicts when installing new phones at an existing site. The multi-protocol IP phone firmware is available at Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome). You can update phones or other IP devices using any of the following:

- **Onboard TFTP Server**: The MiVoice Office 250 now includes a Trivial File Transfer Protocol (TFTP) server to provide firmware downloads to the Mitel 52xx/53xx IP phones, 5610 cordless handset, and 8600, 8620/22, and 8660/62 multi-protocol phones. See “Onboard TFTP Server“ on **page 308**. The Administrative Web Portal interface is the mechanism used to get the firmware to the phones, using the same method as to upgrade system software. Refer to the instructions that accompany the phone-firmware package download.

- **TFTP server**. The Mitel 8690 phones use a TFTP server to update firmware, as in prior releases. The TFTP server allows the phones to pull firmware updates from a TFTP server automatically, rather than push firmware updates to individual phones.
- **Trivial File Transfer Protocol (TFTP) Download**: Included with the IP devices media, or you can download it from Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome) .
- **Upload Utility**: Included with DB Programming.

Onboard TFTP Server

In a Processing Server configuration, the TFTP server resides in the Base Server. The onboard TFTP server is not intended for uses other than TFTP phone firmware downloads.

The TFTP server:

- is not used to upgrade any other MiVoice Office 250 software.
- does not support Mitel 5212/5224 phones at remote locations/teleworker. Mitel 5312/5324 phones are supported.
- supports local and remote downloads. Port 69 must be open on the corporate firewall for phones at remote locations.
- might have issues with some SOHO routers because some of these routers do not provide using custom DHCP options.
- uses ports 69 and 20001 for all transfers to save on socket resources. The server responds on the same port, 69 or 20001, to which the client request is sent. Port 69 must be open on the corporate firewall for phones at remote locations (Mitel 5212/5224 are not supported for tele-worker/remote location use; Mitel 53xx are supported). If left at 0, the system will try port 20001 followed by port 69.

See **Table 53** for information on TFTP, DHCP and other information relevant to Mitel 52xx/53xx phones and Mitel 86xx phones.

Table 53: IP Phone Overview

Setting	Mitel 52xx/53xx Phones		Mitel 86xx Phones
IP Address, Subnet Mask, Default Gateway	DHCP option 43 with the following character string: id:ipphone.mitel.com;sw_tftp=<*MiVoice Office 250 base server IP address*>;call_srv=<*MiVoice Office 250 base server IP address*>;vlan=<*VLANID*>;l2p=<*value*>;dscp=<*value*> (See example on page 246.)	DHCP option 43 or through phone keypad	DHCP or through phone keypad
TFTP server IP address		DHCP option 128 or through phone keypad	DHCP option 66 or through phone keypad. Option 67 must be left blank or "/"
Call control address		DHCP option 129 or through phone keypad	Through UDP discovery[a], .cfg file, or through phone keypad
MITEL IP PHONE		DHCP option 130	n/a
VLAN ID (optional)		DHCP option 132 or through phone keypad	Through phone Web page or .cfg file
Priority on VLAN DHCP validation (required for DHCP)		DHCP option 133 Type the integer and set the priority. If using option 132, you must use option 133.	n/a

a. Because UDP discovery relies on a broadcast, it works only within a Local Area Network (LAN).

IP Phones Supported on the Onboard TFTP Server

The following Mitel 52xx/53xx phones, 5610 cordless handset, and Mitel 86xx phones are supported for downloading firmware from the onboard TFTP server:

- Mitel 52xx/53xx and 5610 (subsequent updates can be initiated through the Admin Web Session pages)
- Mitel 8600, 8620, 8622, 8660, and 8662 (optionally, existing phones can be updated through Admin Web Session pages)

For all other MiVoice Office 250 supported phones, firmware downloads are performed as before 5000 CP v3.0 or as discussed in this document (beginning on page 309). Refer also to the instructions that accompany any phone-firmware-only packages that may be released.

Firmware Management

You must do a software downgrade to revert back to an older firmware version for Mitel phones. Multiple firmware versions are not currently supported; only the latest versions are available for downloading.

Installing Digital Telephones

The following telephones can be installed through the DEI:

- 8528 and 8568 digital telephones
- 8500, 8528, and 8560 digital telephones
- Single-Line Adapters, connected to DEM-16 ports

Before installing the digital telephones, unpack each one and check for damage. If items are damaged or missing, contact Mitel Customer Support. See “Technical Support Contact Information” on **page 1-370**.

NOTICE

The proprietary digital telephones specified in this section are to be installed only on the appropriate MiVoice Office 250 systems.

Each digital telephone from the factory should include the following:

- Baseplate
- One three-pair line cord, 6 ft
- One handset
- One four-conductor coiled handset cord
- Slide-out directory card (*not* available on Mitel 8500, 8520, or 8560)
- Extra feature key templates

To install a digital telephone:

1. Before mounting the modular jack [line jack unit] assembly and connecting the telephone, measure the voltage on the YELLOW terminal of the modular jack [line jack unit] assembly with respect to the BLACK terminal. (The “common” probe of the voltmeter is placed on the BLACK terminal.) It must measure +24 VDC (±2.0 VDC). If a negative voltage is measured, check the cabling for a reversed pair. If tip and ring are reversed, the telephone will not function.

NOTE Unlike the 8520/8560 digital telephones, the newer 8528/8568 telephones use standard RJ11 wiring configuration as defined in TIA 968. See Table 22 on **page 112** for cable termination information.

2. Mount the modular jack [line jack unit] assembly on the wall.
3. Attach the coiled handset cord to the handset and to the handset jack on the bottom of the phone. Place the handset on hook. The bottom of the Mitel 8500, 8520, and 8560 phones have molded channels designed to hold the handset and line cords. For best results when connecting a handset, uncoil a portion of the handset cord and push the cord into the channel so that the flat side of the cord is against the phone.
4. Plug one end of the line cord into the wall-mounted modular jack [line jack unit] assembly. Plug the other end into the line cord jack on the bottom of the phone.

To aid in installation and troubleshooting, display phones show their station circuit number, extension number, and assigned user name for several seconds when power is turned ON and the line cord is first plugged in. This display also appears after a system reset, and whenever the line cord is removed and replaced while power is ON. The station identification displays for several seconds, then the phone changes to the appropriate display, depending on its current status.

Unlike the 8520/8560 phones, the newer 8528/8568 phones use programmable firmware to update the phones. When a phone is powered up, the system checks the firmware file against the phone version. If the phone requires a firmware update due to a new firmware release being available or a system upgrade, the system downloads the updated firmware to the phone automatically. The phone LEDs blink on and off and the LCD remains blank until the phone resets after the firmware update. Depending on the circumstances, the download process can take up to 15 minutes to complete. Do ***not*** unplug the phone until it is operational, which indicates that the firmware update is complete.

5. Perform the self-test for the specific phone model you have, as outlined in the following section.
6. *If necessary*, change the phone ring tone by performing the steps described in the *MiVoice Office 250 Features and Programming Guide*.
7. *If necessary*, replace the phone paper template with one labeled to match the programmed feature buttons.
8. If necessary, adjust the viewing angle. The Mitel 8000 series digital telephones allow you to adjust the viewing angle. If you position the phone viewing angle to 60° or greater, flip the handset hanger to the horizontal position.

Performing the Digital Telephone Self-Test

The self-test operating mode provides a functional test of the phone's hardware. It consists of a series of individual tests transitioned through by pressing and releasing the hook-switch while the LCD provides an indication as to which test is currently being executed.

8528/8568 Digital Telephones

To perform the self-test procedure on an 8528/8568 digital telephone:

1. While pressing and holding the asterisk (*) and pound/hash (#) buttons, unplug and replace the phone line cord. The tests will begin after the asterisk and pound/hash buttons are released.
2. **LCD Test**: Release the buttons. The phone rings momentarily. The LCD shows a dark display—all pixels lit. If all pixels are not lit, contact Mitel Technical Support. The LEDs should not light. If an LED is lit, the button may be stuck. After the first hook-switch transition, the LCD is blank. After the second hook-switch transition, all 0s are displayed. After the third transition, all Xs are displayed. After the fourth transition, all Ms are displayed.

> **NOTE** While the LCD is showing a dark display, you can display hardware and firmware information of a phone for troubleshooting purposes. See “Viewing Hardware and Firmware Information” on page 288 for details.

3. **PKM Test**: If the PKM IM is not present, this step is skipped without any indication to the user. Lift and replace the handset. The phone rings momentarily. If the PKM IM is detected, the LCD shows PKM OK N Keypad, where N is the number of PKMs detected (0-2). This number updates

dynamically if PKMs are attached/removed during this test. It can take up to 2 seconds for the number to update. If this message does not appear, try another PKM IM or contact Mitel Technical Support.

4. **LED Test**: Lift and replace the handset. The phone rings momentarily, and all LED-equipped buttons on the phone and any attached PKMs light. The LCD shows TESTING LED MATRIX. If any of the LEDs do not light, contact Mitel Technical Support.
5. **Keypad Matrix Test**: Lift and replace the handset. The phone rings momentarily. The LCD shows TESTING KEY MATRIX. One at a time, press each of the buttons on the phone and any attached PKMs in any order. If the button is functioning properly, a key tone is heard, the LED lights red, and a specific code is shown on the display. The proper code for each button is diagrammed in Figure 5 on **page 288 and** Figure 6 on **page 289.** If the signal tone is not heard or the displayed code is incorrect, the button is faulty; contact Mitel Technical Support. In addition to testing the keypad on the phone itself, the keys on any attached PKM(s) can also be tested by displaying the hexadecimal key code on the LCD when a key on the PKM is pressed.
6. **Speaker Tones Test**: Lift and replace the handset. The speaker tones are broadcast over the speakerphone speaker. The LCD shows TESTING SPEAKER TONES.
7. **Speaker Volume Test**: Lift and replace the handset. Tones of various volume levels, from softest to loudest, are broadcast over the speakerphone speaker. The LCD shows TESTING SPEAKER VOLUME.
8. **Microphone Test**: Lift and replace the handset. The audio is routed from the hands-free microphone to speaker. The LCD indicates whether the audio path for the microphone test is currently disabled (MICROPHONE OFF) or enabled (MICROPHONE ON). Press a key on the phone to toggle the audio path state.
9. **Handset Tones Test**: Lift the handset, and then press and release the hookswitch. The handset tones are broadcast over the handset receiver. The LCD shows TESTING HANDSET TONES.
10. **Headset Tones Test**: Put on the headset *(if one is attached)* and place the handset back in its cradle. The headset tones are broadcast over the headset. The LCD shows TESTING HEADSET TONES.
11. **Maximum Power Test**: Lift and replace the handset.

WARNING

Possible Hearing Injury. DO NOT hold the handset or headset to your ear when transitioning to the Maximum Power Test. The Maximum Power Test plays a tone out the handset, headset, and hands-free speakers as soon as it begins. This could cause hearing problem due to loud volume.

The maximum power draw test attempts to draw the maximum amount of power from the system by turning on all LCD pixels, all LEDs, and playing a test tone at maximum volume out of the handset, headset, and hands-free speakers. The LCD shows MAX POWER indicating that the maximum power test is in progress by toggling between text and a fully darkened LCD every half-second.

12. **Watchdog Test**: Lift and replace the handset. The phone rings momentarily. The LCD shows TESTING COMPLETED! for several seconds before the phone resets. If the phone watchdog timer fails to reset the phone, the LCD shows WATCHDOG ERROR and eight short bursts of ring tone. Contact Mitel Technical Support.

Viewing Hardware and Firmware Information

During the self-test procedure, you can display hardware and firmware information of an 8528/8568 phone on the LCD. The information can be used for troubleshooting/tracking problems related to specific hardware and software revisions.

To view hardware and firmware information:

While the LCD is showing a dark display during the LCD Test of the self-test procedure (see step 2 on **page 286**), press one of the following buttons:

- Press **1** to display hardware revision information (hardware revision number and hardware ID).
- Press **2** to display firmware revision information (primary and secondary firmware revision numbers).
- Press **3** to display firmware brand (Mitel or Lake).

Figure 5: Mitel 8528/8568 Self-Test Key Matrix

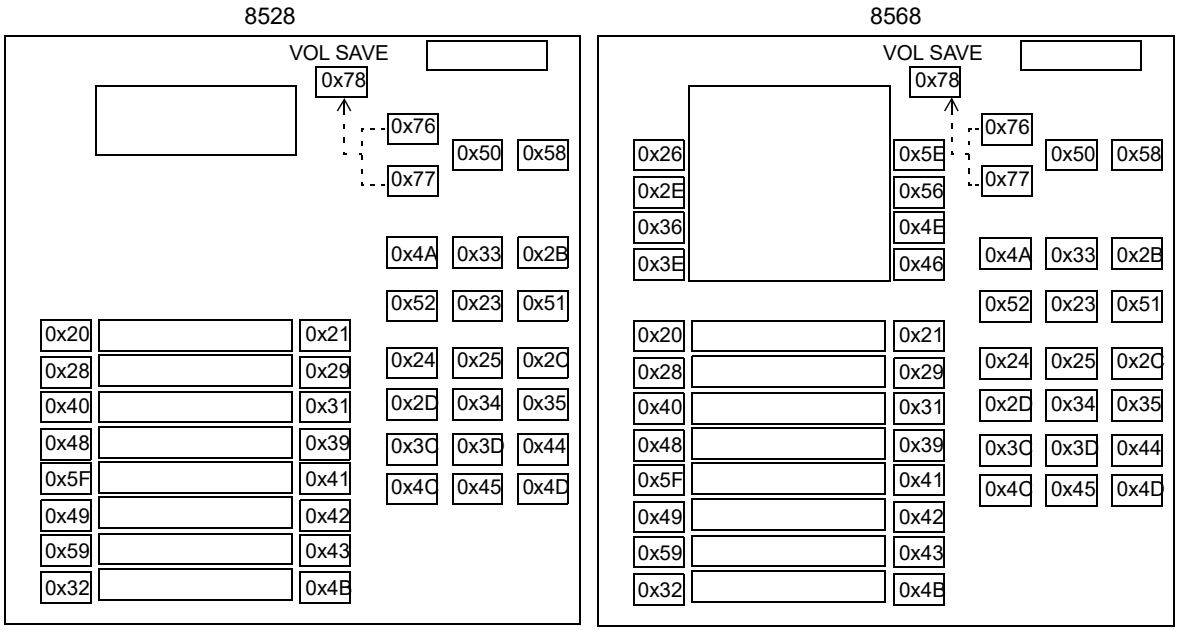
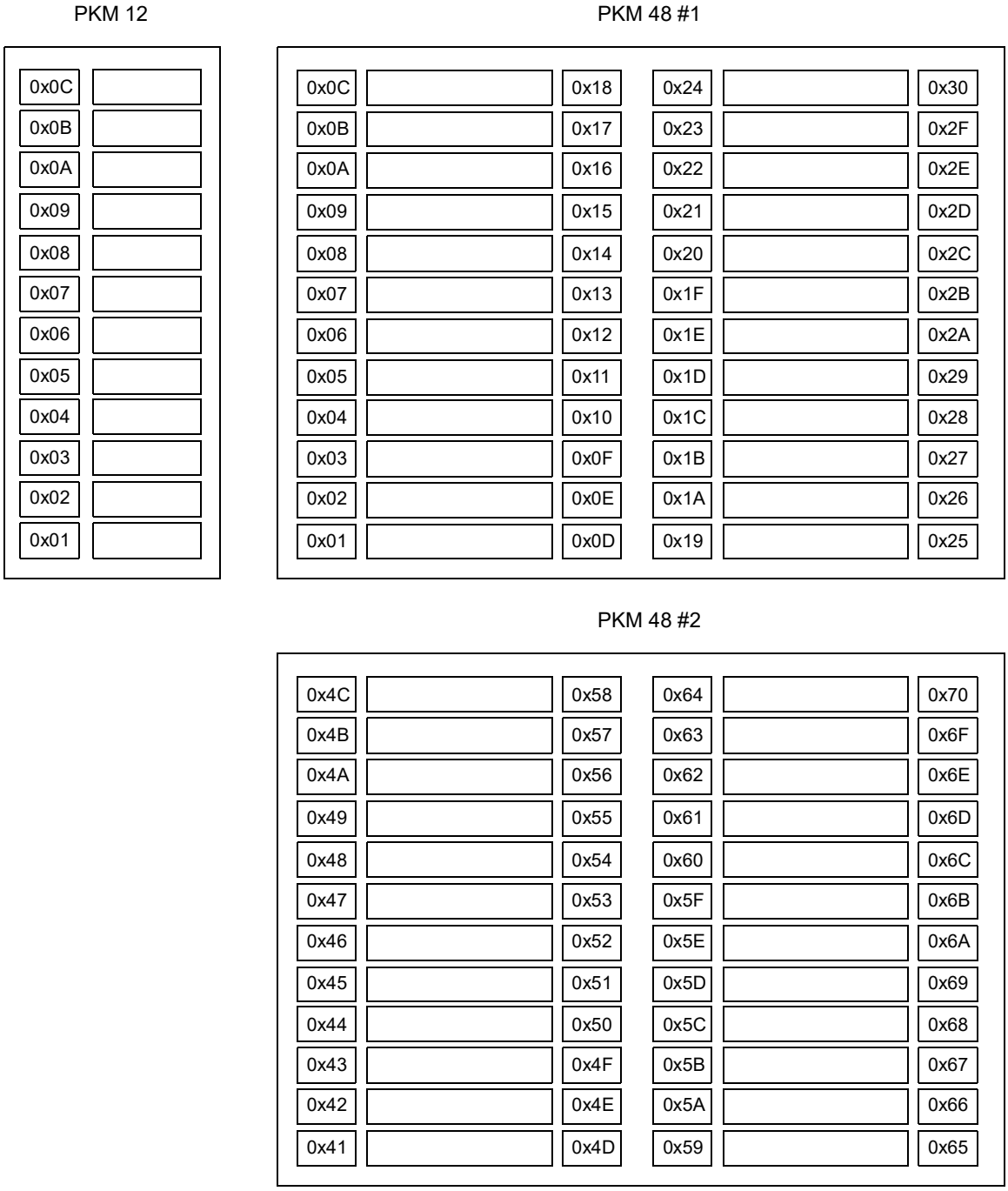


Figure 6: PKM 12 and PKM 48 Self-Test Key Matrix



8500/8520/8560 Digital Telephones

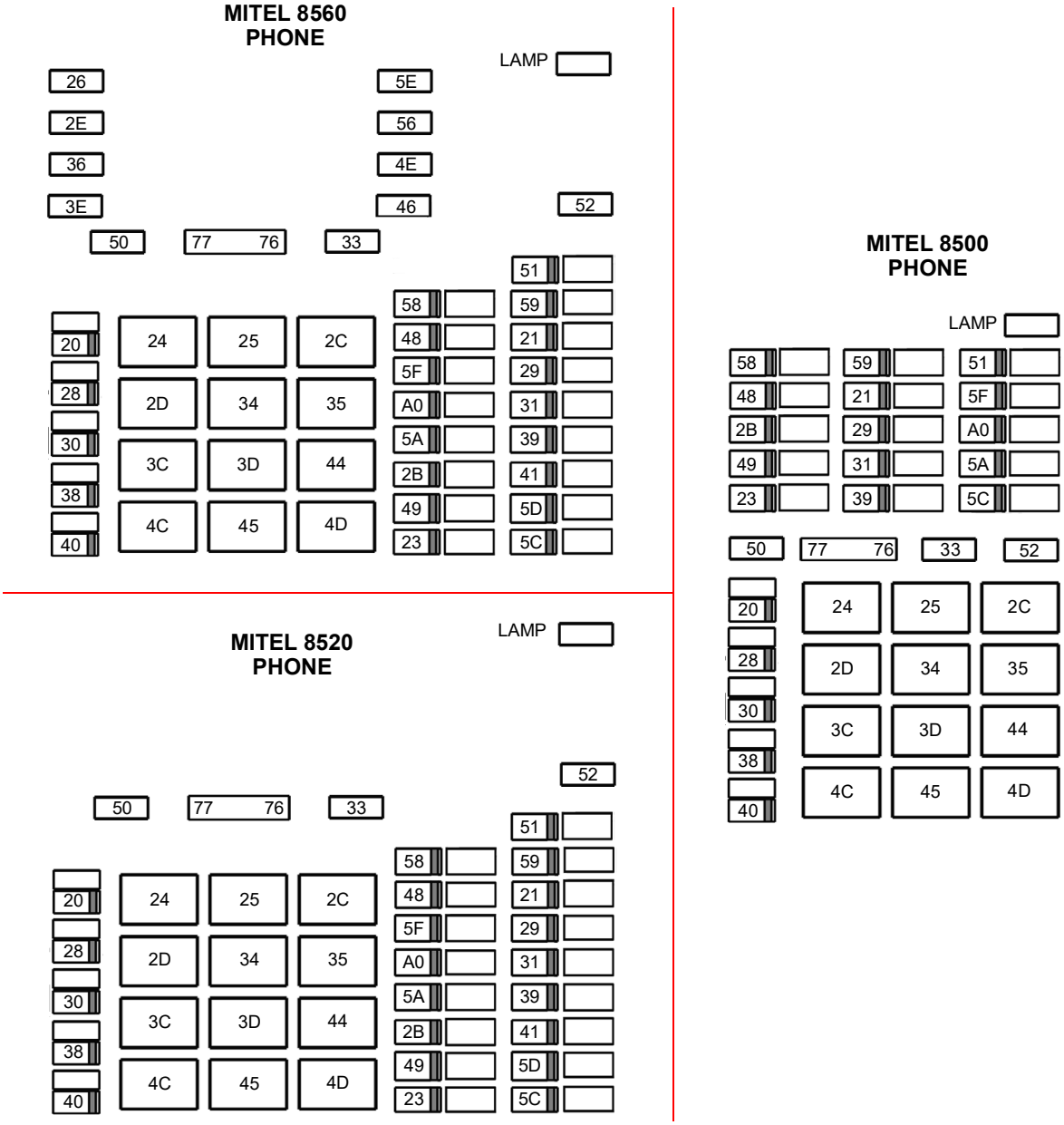
Perform the following procedure for the 8500, 8520, and 8560 digital telephones.

To perform the self-test procedure on 8500, 8520, and 8560 digital telephones:

1. While pressing the asterisk (*) and pound/hash (#) buttons, unplug and replace the phone line cord.
2. Release the buttons. The phone rings momentarily. Display phones show a dark display – all pixels lit. If all pixels are not lit, return the phone. The LEDs should not light. If an LED is lit, the button may be stuck.
3. Lift and replace the handset. The phone sounds an alerting tone. If no tone is heard, return the phone. Display phones show a blank display—no pixels lit. If any pixel is lit, return the phone.
4. Lift and replace the handset. The phone rings momentarily. Display phones show a display filled with Os.
5. Lift and replace the handset. The phone rings momentarily. Display phones show a display filled with Ms.
6. Lift and replace the handset. The phone rings momentarily, and all red, LED-equipped buttons light. Display phones show TESTING LED MATRIX. If any of the LEDs do not light, return the phone for repair.
7. Lift and replace the handset. The phone rings momentarily, and all green, LED-equipped buttons light. Display phones show TESTING LED MATRIX. If any of the LEDs do not light, return the phone for repair.
8. Lift and replace the handset. The phone rings momentarily. Display phones show TESTING KEY MATRIX.
9. One at a time, press each of the buttons on the phone in any order. If the button is functioning properly, a key tone is heard, the LED lights red, and a specific code is shown on the display. The proper code for each button is diagrammed in Figure 7 on **page 291**. If the signal is not heard or the displayed code is incorrect, the button is faulty; return the phone for repair.
10. Lift and replace the handset. The speaker tones are broadcast over the speakerphone speaker. Display phones show TESTING SPEAKER TONES.
11. Lift and replace the handset. Tones of various volume levels, from softest to loudest, are broadcast over the speakerphone speaker. Display phones show TESTING SPEAKER VOLUME.
12. Lift the handset, then press and release the hookswitch. The handset tones are broadcast over the handset receiver. Display phones show TESTING HANDSET TONES.
13. Put on the headset *(if one is attached)* and place the handset back in its cradle. The headset tones are broadcast over the headset. Display phones show TESTING HEADSET TONES.
14. Lift and replace the handset. The phone rings momentarily, and it takes approximately five seconds for the phone to return to normal operation. Display phones show TESTNG COMPLETED! for several seconds.
15. Replace the phone if faulty.

IMPORTANT If the phone watchdog timer fails, the TESTING COMPLETED! display is followed by a WATCHDOG ERROR display and eight short bursts of ring tone. Replace the defective phone.

Figure 7: Mitel 8500, 8520, and 8560 Phones Self-Test Key Matrix



Executive/Professional/Standard/Associate Digital Telephones

Perform the following procedure for the Executive, Professional, Standard, and Associate Display models.

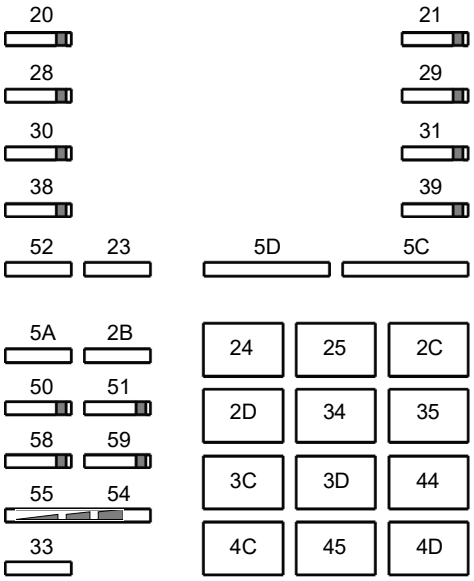
To perform the self-test procedure on Executive, Professional, Standard, Associate, and Basic digital telephones:

1. While pressing the asterisk (*) and pound/hash (#) buttons, unplug and replace the phone line cord.
2. Release the buttons. The phone rings momentarily. Display phones show a dark display—all pixels lit.
3. Lift and replace the handset. The phone rings momentarily. Display phones show a blank display—no pixels lit.
4. Lift and replace the handset. The phone rings momentarily and all LED-equipped buttons light. Display phones show TESTING LED MATRIX. If any of the LEDs do not light, return the phone for repair.
5. Lift and replace the handset. The phone rings momentarily and all of the LEDs go out. Display phones show TESTING KEY MATRIX NN, where NN is a two-digit number.
6. One at a time, press each of the buttons on the phone in any order. If the button is functioning properly, a key tone (or a DTMF tone for Executive and Professional keypad buttons) is heard and, on display phones, a specific code is shown. The correct code for each button appears in Figure 8 on **page 293**. If the signal is not heard or the displayed code is incorrect, the button is faulty, and you should return the phone for repair.
7. Lift and replace the handset. The speaker tones are broadcast over the speakerphone speaker. Display phones show TESTING SPEAKER TONES.
8. Lift and replace the handset. Tones of various volume levels, from softest to loudest, are broadcast over the speakerphone speaker. Display phones show TESTING SPEAKER VOLUME.
9. Lift the handset, and then press and release the hookswitch. The handset tones are broadcast over the handset receiver. Display phones show TESTING HANDSET TONES.
10. Place the handset in the cradle. The phone rings momentarily and it takes approximately five seconds for the phone to return to normal operation. Display phones show TESTING COMPLETED! for several seconds, then, as described in step 4 on **page 285**, the phone identification displays for several seconds.

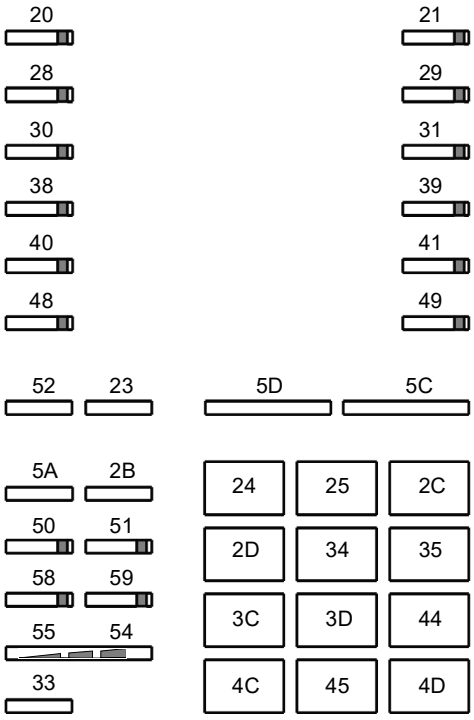
    IMPORTANT If the phone watchdog timer fails, the TESTING COMPLETED! display is followed by a WATCHDOG ERROR display and eight short bursts of ring tone. Replace the defective phone.

11. Replace the phone if faulty.

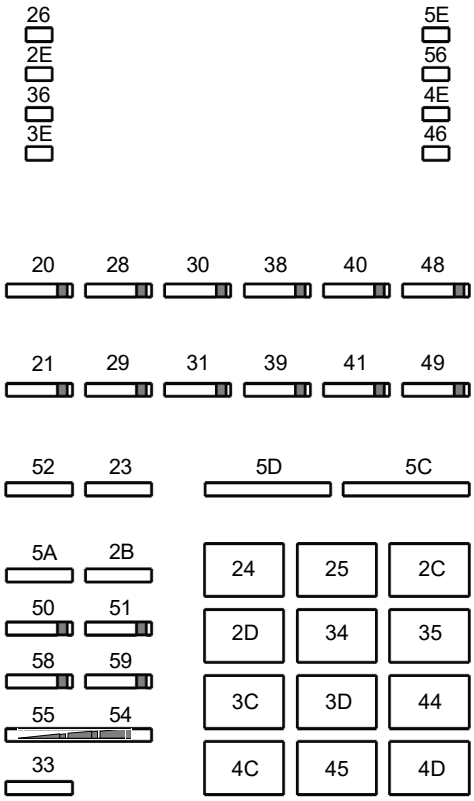
Figure 8: Digital Display and Basic Digital Telephones Self-Test Key Matrix



STANDARD OR ASSOCIATE DISPLAY PHONE



EXECUTIVE OR PROFESSIONAL DISPLAY PHONE



Wall-Mounting 8528/8568 Digital Telephones

CAUTION

Equipment Damage Hazard. Make sure the wall is capable of supporting the weight of the telephone. For additional security, mount the telephone on a plywood backboard.

Select a location away from electrical cables, pipes or other items that could cause a hazard when inserting screws in the wall.

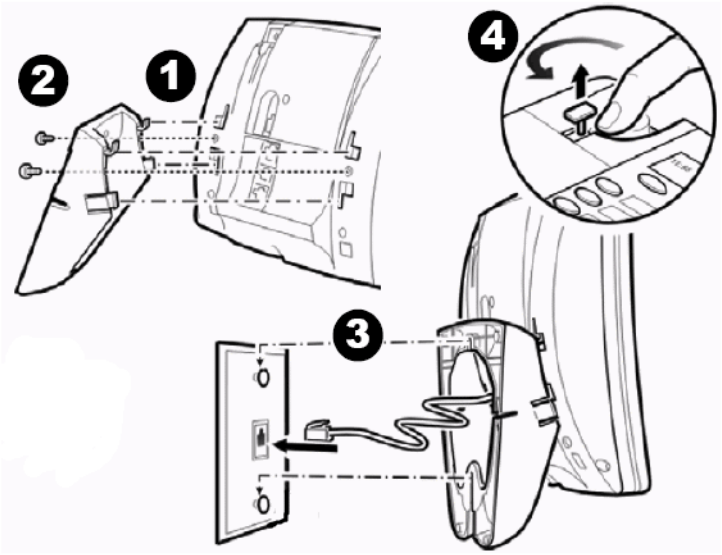
1. Turn the base unit so that the hooks on the flat edge of the base unit bend toward the top edge of the telephone. Hinge the two upper hooks into the top set of notches and snap the two lower hooks into the middle set of notches on the back of the telephone.
2. Secure the base unit using the two screws provided (found in a separate package with the telephone).
3. Do one of the following:
   - Mounting on a wall:

     Insert two mounting screws 3 1/4 inches (8.25 cm) apart. Allow about 3/16 of an inch (0.5 cm) between the wall and screw heads for mounting the telephone. Use #10 screws 1 to 1.5 inches (2.5 to 4 cm) in length for mounting screws and anchoring devices suitable for the wall.
   - Mounting on a wall plate:

   a. Plug the telephone cord into the wall jack.

   b. Place the base unit on the wall plate post, and then push down until the unit is firmly seated.
4. Using your index finger, lift the clip (from the top) located below the hookswitch, turn the clip 180 degrees, and then let the clip snap into place to form a hook. The hook keeps the handset in the cradle when the telephone is wall-mounted.

Figure 3: Wall-Mounting 8528/8568 Digital Telephones



Wall-Mounting 8500/8520/8560 Digital Telephones

Because there are two types of bases used on the Mitel 8000 series digital telephones, two sets of instructions for wall-mounting the phones are included here. Determine which type of base the phone has and follow the appropriate instructions.

For best results when wall-mounting the Mitel 8000 Series digital telephones, Mitel strongly recommends that you use the two plastic rivets shipped with the phone. Once installed, the plastic rivets lock the components into place, providing additional stability for wall-mounted phones.

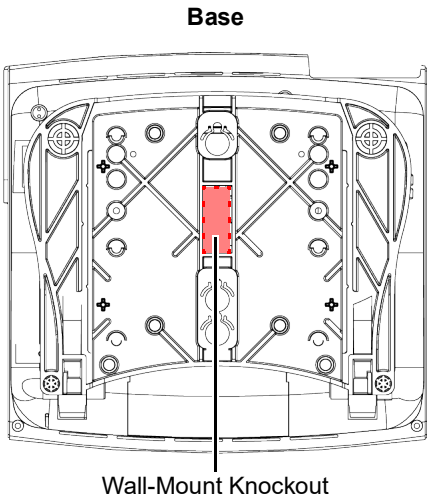
Mitel 8000 series digital telephones are shipped with a plastic adhesive label that covers the rectangular wall-mount knockout on the base of the phone. A portion of the adhesive label must be removed to use the wall-mount knockout.

IMPORTANT	Mitel recommends *not* wall-mounting the Mitel 8000 Series digital telephones if users will be frequently connecting and disconnecting cords (handset, headset, line) and/or cables (Ethernet, USB, power supply). To connect and disconnect these cords and cables from a wall-mounted phone, users will need to remove the phone from the wall.

To wall-mount the Mitel 8000 series digital telephones, use an industry-standard wall-mount bracket (sometimes called a wall jack assembly) and a four-inch line cord. Both of these items are available through the Mitel DataNet CommSource division or Mitel Europe.

To prepare the phone for wall-mounting (regardless of base type):

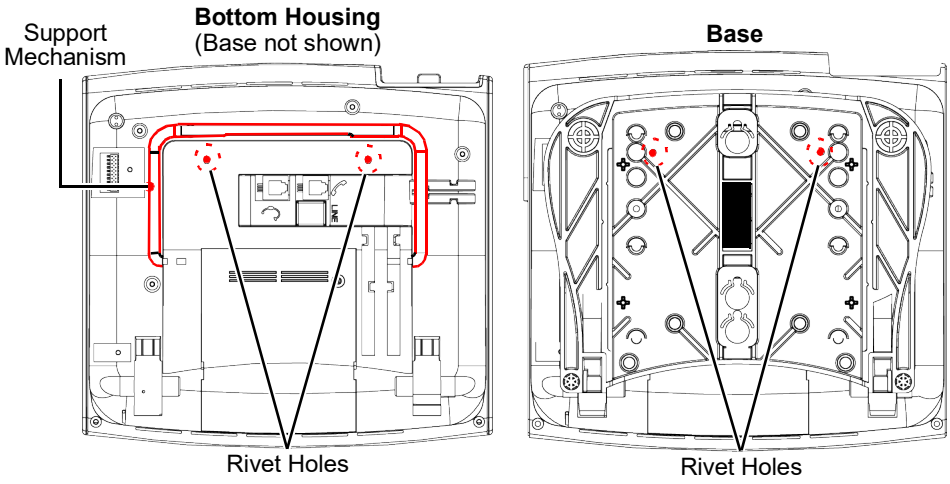
1. Remove the metal plate from the base of the phone by removing the two plastic rivets to unlock it from the back.
2. Slide the cover off. The two plastic rivets are ***not*** included on phones built prior to May 2005.
3. Use a sharp blade to cut out the rectangular portion of the adhesive label covering the wall-mount knock-out (see diagram at right).



The following instructions apply to phones that have a support mechanism (see illustration below) that fits into holes in the base.

To wall-mount Mitel 8000 Series digital telephones having a support mechanism:

1. Collapse the support mechanism completely and secure it against the bottom housing of the phone, as shown below.



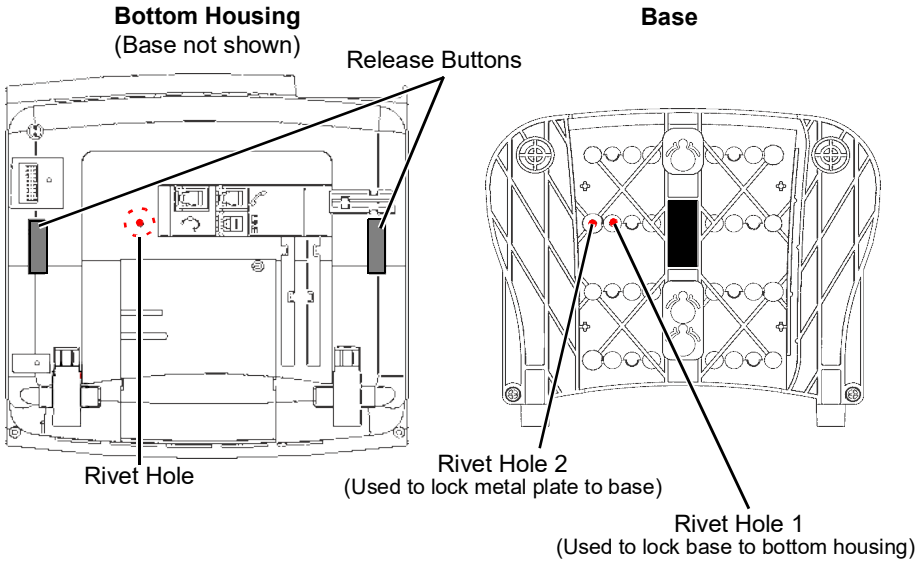
2. Close the base completely so that it lies flat against the phone. Make sure the two rivet holes on the base align with the two rivet holes on the bottom phone housing.
3. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
4. Test the base to verify that it is securely locked to the phone.
5. Insert the two plastic rivets through the metal plate and press them firmly into the holes on the phone base. This locks the metal plate to the base.
6. Re-attach the metal plate to the phone base.
7. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.

8. Flip the handset hanger down to the horizontal position and lock it into place. This will hold the handset in place.

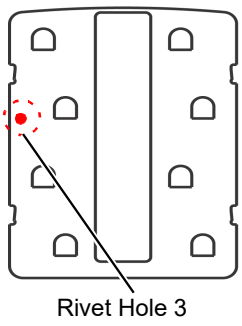
The following instructions apply to phones that have release buttons located on the bottom phone housing. To disengage the locking mechanism in the base, push in both buttons. To lock the base in position, release the buttons.

To wall-mount Mitel 8000 Series phones with release buttons on bottom housing:

1. Close the base completely so that it lies flat against the phone. Rivet hole 1 on the base aligns with the rivet hole on the bottom housing of the phone.



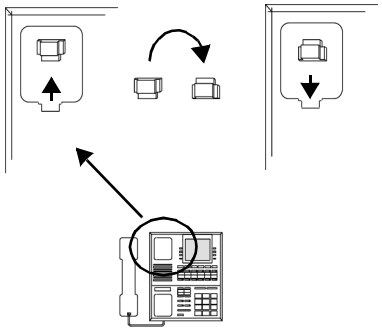
2. Plug the four-inch line cord into the jack on the back of the phone and thread the line cord through the wall-mount knockout on the base.
3. Insert one of the plastic rivets in rivet hole 1 on the base (see illustration above) and press it into the rivet hole on the phone. This locks the base to the phone.
4. Test the base to verify that it is securely locked to the phone.
5. Re-attach the metal plate to the phone base.
6. Insert the remaining plastic rivet through rivet hole 3 (as shown at right) in the metal plate and press it firmly into rivet hole 2 in the base. This locks the metal plate to the base.
7. Plug the line cord into the jack on the wall-mount bracket and position the phone securely on the wall-mount bracket.
8. Flip the handset hanger down to the horizontal position and lock it into place. This will hold the handset in place.



Wall-Mounting Executive/Professional/Standard/Associate Digital Telephones

To mount the phone on a wall, use an industry-standard wall-mount bracket (sometimes called a wall jack assembly). If a wall-mount bracket is not available, use the two-screw method outlined below:

1. Remove the phone baseplate from the top half of the back of the phone.
2. Rotate the baseplate 180° and attach it to the bottom half of the back of the phone. If a PCDPM is already installed inside the baseplate, first open the phone and reroute the PCDPM-to-keyset interface cable through the knockout in the bottom half of the phone. See Figure 13 on **page 356**.
3. Using a ruler, mark the location of the phone mounting holes on the wall. The centers of the two perpendicular mounting holes on the back of the phone are four inches apart.
4. Drive a screw into the center of each mounting hole marking. Allow the head of the screw to protrude approximately ¼ inch.
5. To hold the handset in place while the phone is wall-mounted, slide the reversible cradle hook out, rotate it 180°, and reinsert it as shown below.



6. Position the phone mounting holes over the screws and slide the phone into position on the wall. If necessary, adjust the screws to ensure the phone is held firmly in place.

Adjusting the Baseplate on Executive/Professional/Standard/Associate Digital Telephones

The Professional, Associate, and Basic phones have adjustable baseplates that allow users to set the angle of the phone.

To change the baseplate position:

1. Gently press on one edge of the baseplate and lift it off of the phone.
2. If you have a PCDPM installed, route the cables so that they will be between the baseplate and the bottom of the phone.
3. Position the baseplate over one of the three sets of holes and snap it into place. The curved edge should be toward the top of the keyset.

Installing an Optional Headset on a Digital Telephone

For headset requirements, see “Optional Headsets” on page 101.

To attach a headset to an applicable digital telephone*:

*Includes Executive, Professional, Standard, or Associate Display phones; or the Mitel 8500 phone.

1. Remove the coiled handset cord from the handset jack on the bottom of the phone. Leave the handset in the cradle.
2. Insert the headset modular plug into the jack. Plug in the headset power source, if used.
3. If the headset has an ON/OFF switch, turn ON the headset.
4. On the phone, press the Special button and enter the Headset Enable feature code (default value is 315).
5. *To disable the headset*, press the Special button and enter the Headset Disable feature code (default value is 316). Unplug the headset and reconnect the handset.

To attach a headset to a Mitel 8520 or 8560 phone:

1. Insert the headset modular plug into the jack located underneath the phone. Plug in the headset power source, if used.
2. If the headset has an ON/OFF switch, turn ON the headset.
3. On the phone, press ∞ and enter the Headset Enable feature code (default value is 315).
4. *To disable the headset*, press ∞ and enter the Headset Disable feature code (default value is 316). Unplug the headset and reconnect the handset.

Installing Single Line Phones

AC—or tip/ring-controlled—single line phones can be installed using SLM-4 or SLM-8 modules, SLA or IP SLA units, the Data Device port on an MDPM attached to a digital telephone, or the two-port single line jack (RJ-14) built-in to the back of the CS Controller. Program the items discussed in these sections using information discussed in the Programming chapter of the *MiVoice Office 250 Features and Programming Guide*.

Single Line Modules

If required, install an SLM-4 in the Mitel controller in accordance with the standardized procedure, “Installing a Four-Port Single Line Module (SLM-4)” on **page 1-199**. As applicable, program the phone in accordance with the procedure on programming a four-port single line module in the *MiVoice Office 250 Features and Programming Guide*.

If required, install a SLM-8 module in the Digital Expansion Interface chassis in accordance with the procedure, “Installing an Eight-Port Single Line Module (SLM-8)” on **page 1-230**. As applicable, program the phone in accordance with the procedure on programming an eight-port single line module in the *MiVoice Office 250 Features and Programming Guide*.

Single-Line Adapter (SLA)

If required, install an SLA in accordance with the procedure appearing at “Installing an SLA” on **page 1-300**.

IP Single-Line Adapter (IP SLA)

If required, install an IP SLA in accordance with the procedure at “Installing an IP SLA” on **page 1-304**.

Modem Data Port Module (MDPM)

If required, install a PCDPM in a digital telephone in accordance with “Installing a PCDPM” on **page 1-346**. If required, connect an MDPM to the PCDPM in accordance with “Connecting PCDPM Cables” on **page 1-357**.

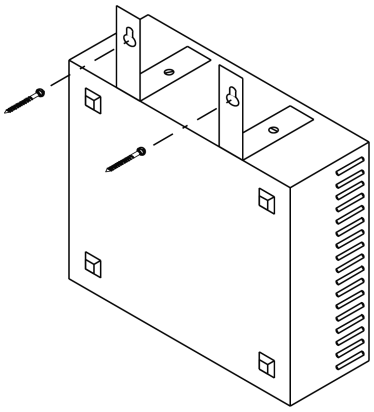
Installing an SLA

When installed in a DEM-16 port, each SLA unit can support one DID [DDI] trunk, two single line phones, or one off-premises extension (OPX).

Wall-Mounting an SLA

The SLA can be placed horizontally on a table, or it can be wall-mounted using the two L-shaped brackets that accompany the unit.

To wall-mount an SLA unit:



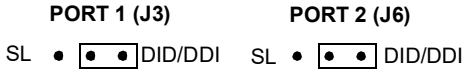
1. Remove the two screws on one side of the unit and use them to attach the two L-shaped wall-mounting brackets. See the figure at right.
2. Position the SLA on the MDF backboard and mark the location of the mounting screw holes on the wall. Set the SLA aside.
3. Drive a screw into the center of each mounting hole marking, allowing the heads of the screws to protrude slightly. Use screws of sufficient strength to support the SLA.
4. Hang the SLA on the screws. If necessary, adjust the screws to ensure the SLA is held firmly in place.

Setting Hybrid Balance Straps for a DID [DDI] Trunk on an SLA

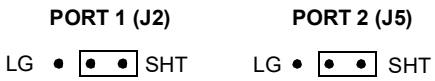
SLAs are shipped with both ports set for single line device operation.

To change the port type if connecting a DID (DDI) trunk:

1. Remove the cover and set the corresponding port for DID (DDI) operation.



2. If necessary, adjust the hybrid balance setting from short (<.6 mi, or 1km) to long (>.6 mi, or 1km).



3. Replace the cover.

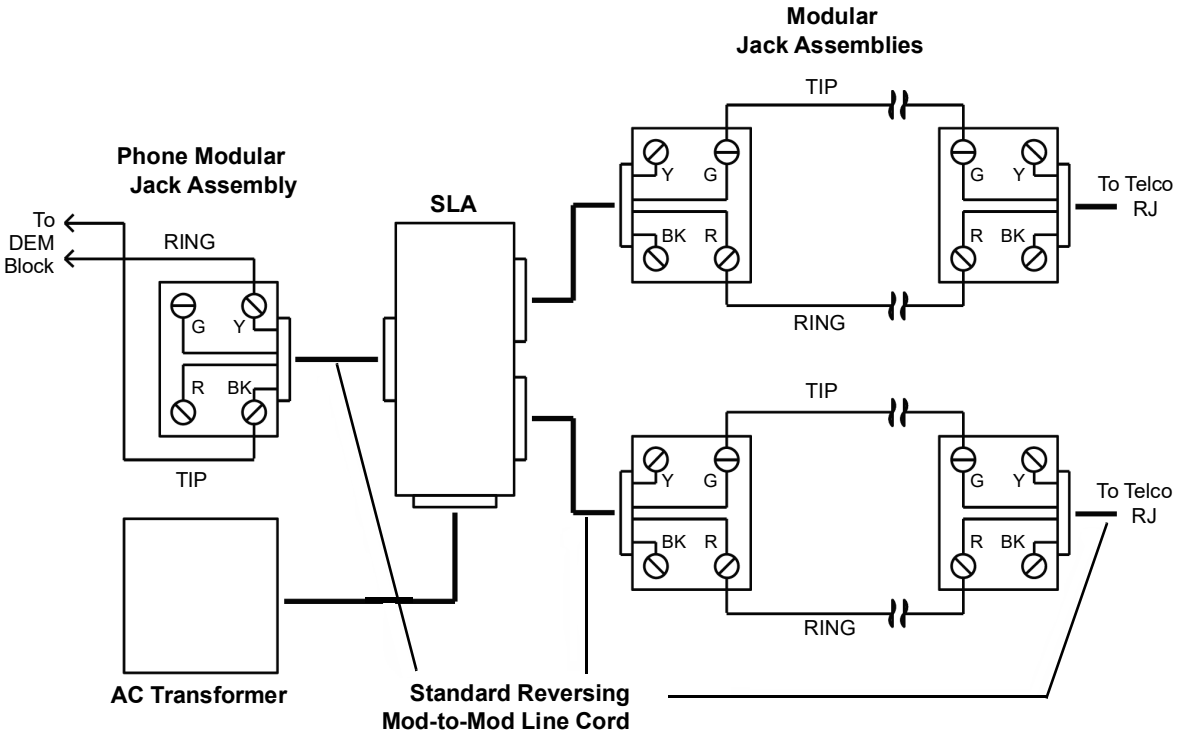
Installing DID Trunks on an SLA (USA Only)

Connect the Direct Inward Dialing (DID) trunks to the Single-Line Adapters as outlined below. See Figure 3 on **page 302**. DID trunks are also supported through T1 circuits. In addition to DID trunks, SLAs can also be used to install single line phones, playback devices, and off-premises extension (OPX) stations. Any combination using the two ports on the SLA is possible. For a diagram, see Figure 4 on **page 303**.

1. Mount one four-conductor modular jack assembly next to each telephone company RJ jack.
2. For each modular jack assembly mounted in step **1**, mount a corresponding four-conductor modular jack assembly near the SLA location.
3. To connect the modular jack assemblies mounted in steps 1 and 2:
   - **a.** Run one-pair cable between the corresponding modular jack assemblies.
   - **b.** Wire each end of the one-pair cables onto their respective modular jack assemblies. For a wiring diagram, see Figure 3 on **page 302**.

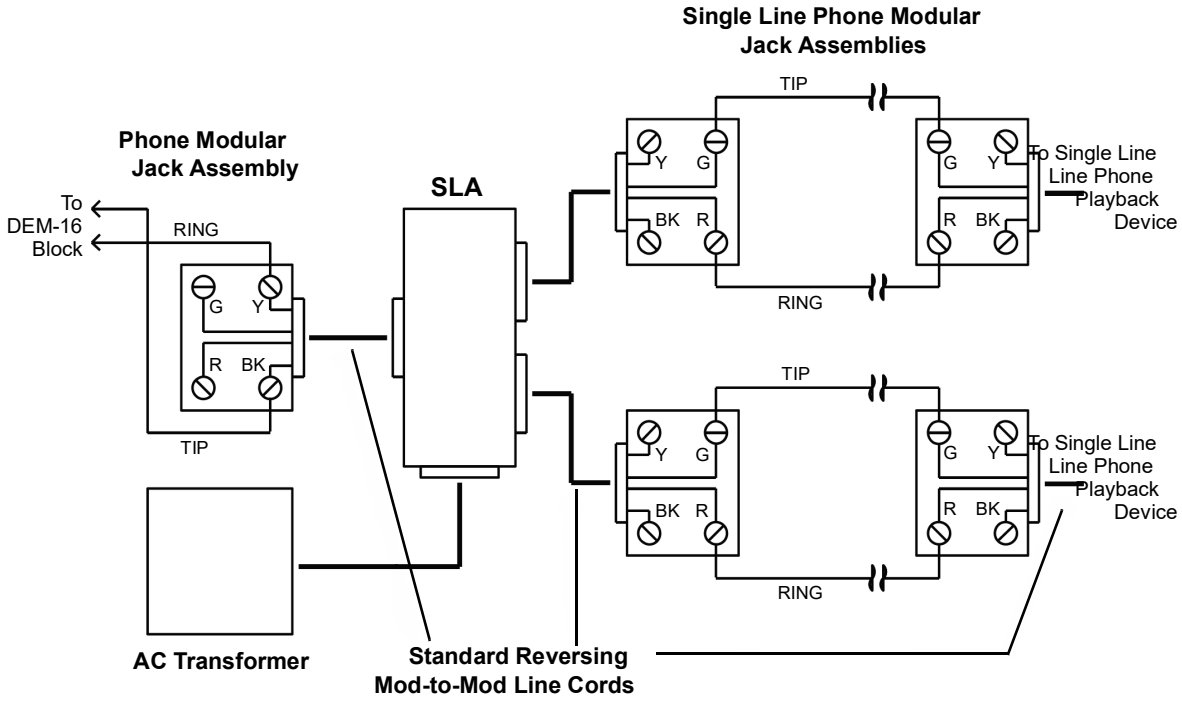
4. Plug one end of a two-pair, mod-to-mod line cord into each RJ jack and plug the other end into the corresponding modular jack assembly beside it.
5. At the MDF backboard, plug one end of a two-pair mod-to-mod line cord into each modular jack assembly. After programming the DEM-16 port, insert the plug into the Circuit 1 or Circuit 2 jack on the SLA, as applicable. See “Programming an SLA for Dual DID Trunks” on **page 1-303**.

Figure 3: SLA Terminations for DID Trunks (USA Only)



If a single line phone is located near the SLA, it can be plugged directly into the unit.

Figure 4: SLA Terminations for Single Line Phones with a U.S. System



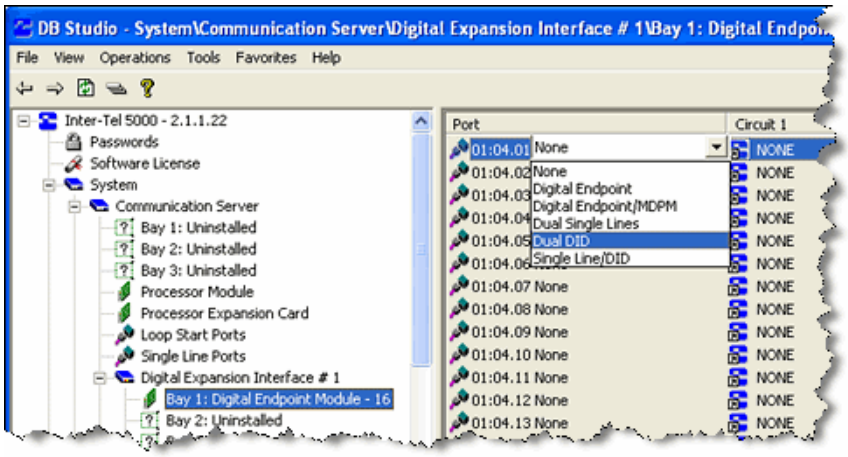
If a single line phone is located near the SLA, it can be plugged directly into the unit.

Programming an SLA for Dual DID Trunks

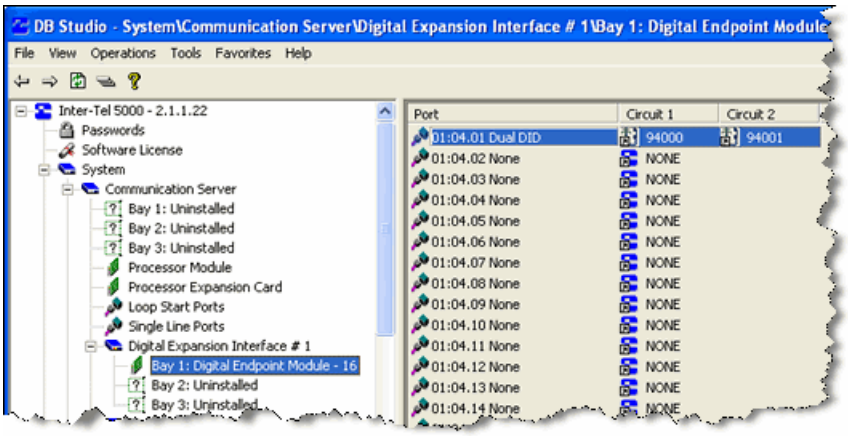
You can use an SLA to install DID trunks on the MiVoice Office 250. The trunks coming in from the CO [local exchange] attach to the SLA that has been programmed for Dual DID functionality on a DEM-16 port.

To program an SLA for dual DID trunks:

1. Open a session in the System Administration & Diagnostics application.
2. From the DB Programming screen, select System – Communication Server – Digital Expansion Interface # <*n*> – **Bay <*n*>:** that is equipped with a Digital Endpoint Module - 16.
3. In the right-side pane, click an available Port and from the drop-down menu click **Dual DID**. The Create DID Extension window displays (not shown).



4. In the Create DID Extension window (not shown), select the Starting Extension trunk number and click **OK**.
5. Circuit 1 and Circuit 2 columns appear, containing the selected extension trunk numbers.



6. Connect a DID trunk into the appropriate jack on an SLA installed in accordance with “Installing DID Trunks on an SLA (USA Only)” on **page 1-301**.

Installing an IP SLA

The IP SLA provides an interface between a standard single line phone and a 10/100Base-T network. The MiVoice Office 250 operates *only* on a 100Base-T network. The IP SLA allows you to make calls through the IP network. The IP SLA operates like a standard SLA. However, its interface to the system is replaced by the Ethernet IEEE 802.3 10 Mbps UTP interface. The IP SLA uses flash memory and its software can be updated over the LAN.

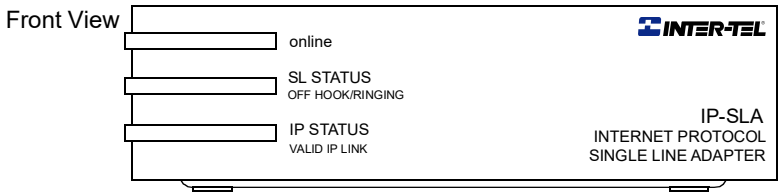
IMPORTANT	Fax over IP (FoIP) and modems are not supported through an IP SLA. To connect a fax machine, attach it to a single line port on one of the single line interfaces supported by the MiVoice Office 250. Mitel currently supports T.38 FoIP only.

To install IP SLAs and associated single line phones:

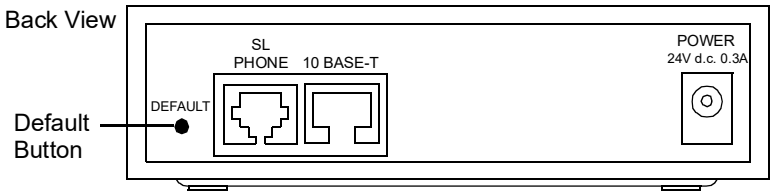
1. Using CAT5e or CAT5 cable, connect the 10BASE-T jack on the IP SLA to the KS/SLA jack on the 24 VDC adapter. For the correct power brick, see “Replacement Parts” on **page 1-371**.
2. Using another CAT5e or CAT5 cable, connect the LAN jack on the 24 VDC adapter to the LAN.
3. Connect the 24 VDC adapter to a 120 V wall socket.

> **NOTICE**
>
> **Possible Equipment Damage.** Use only the appropriate power adapter with the IP SLA. Do ***not*** attach any device (for example, hubs) between the KS/SLA jack on the adapter and the 10BASE-T jack on the IP SLA because power is supplied through the cable.

4. Connect the single line cord to the **SL PHONE** jack. Within a few moments, the single line phone will receive dial tone from the system.
5. Check connection state information while the IP SLA is connecting to the IPR application. The following LEDs show connection state information, as described below.
   - **online**: Indicates a connection between the IP SLA and system call processing.
   - **SL STATUS**: Flashes in time with ringing voltage and is lit when the phone is off-hook.
   - **IP STATUS**: Lights when there is a valid Ethernet connection to the device.



The DEFAULT button has been added to the IP SLA (revision level -AC or later). This button allows you to perform the two functions described on the following page.



Defaulting IP SLA Settings

This allows you to change the IP SLA configuration (such as an IP address) to its default settings.

To default the IP SLA:

1. Press and hold the DEFAULT button using a pinhole tool (such as a bent paper clip). The SL STATUS LED flutters for five or six seconds. The default button is not a reset button.
2. When the online LED is lit, remove the pinhole tool from the DEFAULT button. The online LED turns off.

Placing the IP SLA in Download Mode

To place the IP SLA in download mode:

1. Power down the IP SLA by either removing the cable from the power jack, if present, or by removing the cable from the 10BASE-T jack.
2. Press and hold the DEFAULT button using a pinhole tool while reconnecting power to the IP SLA. The online and SL STATUS LEDs start flashing alternately.
3. Remove the pinhole tool from the DEFAULT button. The IP SLA is now in the download mode.

To change the mode using an older IP SLA (revision level earlier than –AC), remove and reconnect the cable from/into the 10BASE-T jack six times within three seconds. The online and SL STATUS LEDs start flashing alternately. If the LEDs do not flash alternately, repeat this procedure until the LEDs start flashing.

SYSTEM UPGRADES AND EXPANSION

This portion of the manual provides procedures for upgrading and/or expanding MiVoice Office 250 systems. To upgrade to the latest software release, see “Network Environment Requirements” on page 307.

An upgrade and/or expansion may involve one of the following scenarios:

- Increase only the system software version release number, such as going from v3.2 Release 2 to v3.2 Release 3.
- Increase the system software version, such as going from v3.2 to v6.0.
- Expand resources by adding a PEC-1 and/or PS-1, while maintaining the same software version.
- Expand resources and increase the software release number or version.

Guidelines for Expanding or Upgrading a System

When adding hardware to or upgrading the version of a system, observe the following guidelines. For detailed instructions and figures for each step, see the appropriate sections in this chapter.

1. Advise the customer and plan to do the upgrade/expansion when it has the least impact on their operation.
2. If the changes affect the Processing Module (PM-1), Processor Expansion Card (PEC-1), Processing Server (PS-1), or customer database, perform a complete save of the customer database. For details, refer to the database save and restore information in the *MiVoice Office 250 Features and Programming Guide*.
3. If external voice processing will be affected, save the voice processing database and prompts. If using Unified Voice Messaging only, back up the voice processing database to a USB flash drive.
4. If performing an upgrade, follow the instructions provided with the software for saving, converting, and restoring software. See “Upgrading Existing Systems to v5.x and later Software“ on **page 311** for information.
5. Make sure the software license includes the appropriate features required for the installation or upgrade. If the license does not match DB Programming, an error message appears on the LCD panel and administrator phone. For additional information, refer to the software license information in the *MiVoice Office 250 Features and Programming Guide*.
6. Refer to the Programming chapter in the *MiVoice Office 250 Features and Programming Guide*, for procedural guidance to restore the database or voice mail database and configure and program the additions to the system.
7. Train end users on any new features or options that have been installed. Provide them contact information for directing questions or issues.

Network Environment Requirements

For the most efficient network connectivity and system performance, the MiVoice Office 250 must operate in a suitable network environment. Detailed characteristics appear in *VoIP Data Network Requirements*, part number 835.2885.

If upgrading a Base Server equipped with a PS-1, both units:

- Must be connected to the same local area network (LAN).
- Must be on the same subnet.
- Must be on the same Virtual LAN (VLAN).
- Must be connected on a 100 Mbps or faster switch (configured to auto-negotiate both speed and duplex mode).
- Cannot have a Network Address Translation (NAT) device between them.

If a Base Server—as well as a PS-1—resides behind a firewall and you want to make changes to the system from a remote location, make sure the system is accessible. To gain access to DB Programming, IP phones, Secure Shell (SSH), Diagnostics Monitor, or Administrative Web Sessions, the network administrator must open a specific range of User Datagram Protocol (UDP) ports on the firewall.

Table 54 shows the UDP port ranges applicable to v3.x. These port assignments apply to the Processor Module (PM-1) and the Processor Expansion Card (PEC-1) in the CS Controller. These port assignments do ***not*** apply to the Processing Server (PS-1). All DB Programming occurs through the Base Server, unless the site is installing a PS-1.

Table 54: UDP Port Assignments for v3.x or later

System Version	UDP Port Range
v3.x or later	UDP 6004–7039 (RTP and RTCP) Processor Module (PM-1) base port – 6004 Processor Expansion Card (PEC-1) base port – 6604

The MiVoice Office 250 does not support IP networking through a Network Address Translation (NAT) device.

Onboard TFTP Server

In a system equipped with a PS-1, the TFTP server resides in the Base Server. The onboard TFTP server is not intended for uses other than TFTP phone firmware downloads. See page 283 for additional information.

See Table 53, “IP Phone Overview,” on **page 283** for information on TFTP, DHCP and other information relevant to Mitel 52xx/53xx phones and Mitel 86xx phones.

IP Phones Supported on the Onboard TFTP Server

The following Mitel 52xx/53xx, 85xx, and 86xx phones are supported for downloading firmware from the onboard TFTP server: Updates can be initiated through the Admin Web Session pages.

- Mitel 5212, 5224, 5304, 5312, 5324, 5320, 5330, 5340, 5360, and 5610
- MItel 8528 and 8568
- MItel 8600, 8620, 8622, 8662

For all other MiVoice Office 250 supported phones, firmware downloads are performed as before 5000 CP v2.4 or as discussed in this document (beginning on page 309). In addition, the Phone

Firmware Upgrade Instructions document, part number, provides information on updating IP phone firmware for the following models:

Firmware Management

You must do a software downgrade to revert back to an older firmware version for Mitel phones. Multiple firmware versions are not currently supported; only the latest versions are available for downloading.

Software Packaging

Phone firmware images are maintained on the MiVoice Office 250 server for each of the supported Mitel IP phones. The phone firmware is maintained in packages to be upgraded as part of the MiVoice Office 250 upgrade. The images must be explicitly installed.

Only one release of each firmware is maintained on the MiVoice Office 250 server.

Package Mechanism

The standard MiVoice Office 250 packaging procedure is used. There are two packages:

- **cs-mitl-ep**: classic Mitel IP phones package (automatically installed)
- **cs-ep-mitl-85x8:** newer digital telephones (automatically installed)

During the MiVoice Office 250 upgrade (selected from the AWP System Upgrade Web page), the installed packages are displayed along with the other system packages that are installed during the MiVoice Office 250 upgrade.

On a system upgrade from an earlier release (for example, 5000 CP Release 3.1) not containing phone firmware, the Mitel 86xx phone firmware packages are not automatically installed, but the Mitel 52xx/53xx phone firmware IS automatically installed.

For any MiVoice Office 250 software release, the latest phone packages will always be delivered, even if they do not match the version numbers. For example it will not be uncommon for a 5000 CP v3.0 release to contain 5000 v3.1 phone packages. The phone firmware is not system software specific. The new phone firmware will work with older software. The only exception is if the phone type is not supported on the older or newer system software.

Package Downgrade

System downgrades are supported. For a system downgrade to an earlier release (for example, 5000 CP Release 2.3) not containing phone firmware support, the phone firmware packages and contents are removed from the 5000 CP/MiVoice Office 250.

Space Management

Before any package upgrade starts, compact flash available space checks are performed on the MiVoice Office 250 as follows:

- A check for available space is performed before package transfer starts.
- A check for available space is performed after all packages are downloaded to the MiVoice Office 250 and before the package contents are upgraded.

If the system upgrade determines there is not enough compact flash space available to complete the system upgrade, the system upgrade displays a message informing you of a lack of available space, and it will not install any packages. You are then prompted to remove some voice-mail messages and try again. The system upgrade aborts.

Upgrading Existing Systems to v5.x and later Software

This section provides detailed information on upgrading an existing v5.x (or earlier) or 6.0 system to the v6.2 or later system software release.Version 6.2 or later software operates on the HX Controller hardware platform.

Note: Version 6.2 software is not supported on the CS Controller hardware platform.

Please read all of the following information before continuing.

Two Upgrade Options

There are two available upgrade options:

- **Replace the Memory Card:** The first option (recommended) is to simply replace an existing v1.2 or later compact flash-type memory card with a new v6.2GA compact flash-type memory card (one GB or two GB) that ships with the v6.2 GA software already installed on it. Note that v6.2 GA compact flash-type memory cards also include the latest core Linux file system that was released with the v4.0/v5.x system software release. See page 372 for a list of available v6.1 or later compact flash-type memory cards. This procedure is documented beginning on page 313.
- **Upgrade the Memory Card:** If a v6.2 GA compact flash-type memory card is not available, the second option is to upgrade an existing v4.0 or v5.x system via the Administrative Web Portal (AWP) application. Note that this ***requires at least a one GB card***.

  *If upgrading a **v3.2** or earlier system with a minimum CF size of 1 GB*, the system must first be upgraded to v4.0 or v5.x using a two-step method that requires an intermediate core Linux file system package be installed prior to installing the v4.0 or v5.x software version release package. This procedure is documented beginning on page 315.

  *If upgrading a **v4.0** system,* the latest core Linux file system was already installed along with v4.0, so only the system software is upgraded. This procedure is documented beginning on page 316.

> **NOTE** Once the core Linux files have been downloaded and upgraded for either v4.0 or v5.x GA, there is no need to download them again for future v4.x or v5.x software upgrades.

If the system is equipped with a Processing Server (PS-1), the PS-1 is upgraded separately using a boot CD (for v3.2, v4.0, v5.x, v6.0, or v6.1) or the AWP application (for v4.0, v5.x, v6.0, or v6.1). For a v3.2 or earlier PS-1, you must use the boot CD in order to upgrade the Linux file system that was changed in v4.0 (you cannot upgrade the Linux file system using the AWP application). Specific instructions are included in the procedures documented on the following pages.

Compatibility

If replacing an existing memory card with a new v6.2 GA card, Mitel supports software upgrades from v1.2 or later, as long as the appropriate licensing progression is obtained.

If upgrading an existing memory card to v6.1 GA, Mitel only supports upgrades from v4.0 or v5.x to v6.2 GA. Systems with software v3.2 or earlier must first upgrade to v4.0 or v5.x prior to upgrading to v6.2 GA. (Note that older databases themselves can be converted directly to v6.1 GA.)

Software downgrades from v6.2 GA to v4.x or v5.x are supported. However, software downgrades to v3.x or earlier are not supported. To revert to a v3.x system, remove the v6.2 GA compact flash-type memory card, install a v3.x compact flash-type memory card, and then restore a v3.x database.

Once an existing system is upgraded to v4.0, v5.x, v6.0, or v6.2GA, its database and compact flash-type memory card are interchangeable between CS Controller hardware and HX Controller hardware.

Enterprise Messaging is not compatible with 5000 v6.0 or later software.

If using Customer Service Manager (CSM) as the system's customer interaction solution, upgrade CSM to v5.1 Service Pack 4 (or later) in order to ensure full compatibility with the 5000 v6.0 Meet-Me Conferencing and OfficeLink Assistant features. Refer to the latest CSM documentation for upgrade details.

Storage and Space Limitations

NOTICE

Version 6.1 requires a minimum one GB compact flash-type memory card. The 512 MB card is not supported in v4.0 or later.

If installing voice prompts, MOH files, or using UM Forward and Copy, keep in mind space limitations on the compact flash-type memory card. To maximize memory, install only the necessary language prompt package(s), and remove all others.

When upgrading, the system checks to make sure there is enough available free space (about 125 MB) on the compact flash-type memory card. If there is not enough space to complete the upgrade for all packages, a dialog box appears warning you, and the upgrade process is stopped. Delete unneeded voice mail messages, languages, MOH files, etc. to free up space, or upgrade to a larger compact flash-type memory card, and then restart the system upgrade.

If upgrading an existing compact flash-type memory card (upgrade option two described on the previous page), note that the largest package in the "core_upgrade" set of packages is approximately 25 MB. Be sure to use a TFTP server that can handle larger files.

UVM (BVM) Voice Data Storage

There have been some changes to the voice data capacity limits on the Compact Flash. This was necessary to allow appropriate space for the v5.0 system software and allow for future functionality. To ensure that you have appropriate space, look at the table below.

Note that the compact flash size does not increase the voice mail storage on a PS-1 since Unified Voice Messaging uses the hard drive to store the voice mail data. See **Table 55** below.

Table 55:

	Compact Flash vs. Voice Mail Storage		
Compact Flash Size	Voice Mail Storage (Non PS-1) G.729 (default)	Voice Mail Storage (Non PS-1) G.726	Voice Mail Storage (PS-1) G.711
1 GB	6,000 minutes (100 hours)	1,500 minutes (25 hours)	30,000 minutes (500 hours)

Table 55:

	Compact Flash vs. Voice Mail Storage		
2 GB	18,000 minutes (300 hours)	4,500 minutes (75 hours)	30,000 minutes (500 hours)

System Restarts

System restarts (reboots) are a normal part of the upgrade process and may take longer to complete than normal restarts. Do not interrupt any system operations during a system restart. If needed, you can monitor the progress of the upgrade process via the system's USB-B port and a terminal emulation program.

Other Considerations

Consider the following when upgrading to v6.0 or later. Refer to the main installation manual for any information not discussed in this document.

- **Hardware and Software Options**: Upgrading to v6.0 or later can possibly involve several combinations of hardware and software depending on which version is being upgraded and which hardware options are being added.
- **Upgrade Licensing**: A USB key or serialized compact flash-type card holds the MiVoice Office 250 serial number for a system being upgraded. Licensing is uploaded from DB Programming. The license file, for either the USB key or the compact flash-type card, must match the serial number on the USB key or the compact flash-type card or the upload will fail.
- **Browser Requirements:** Use one of the latest supported browsers:
  - Microsoft Internet Explorer® (IE) 7 and 8, and from v6.0 on, IE 8, 9, 10
  - Mozilla® Firefox® 4 and 5, and from v6.0 on, Firefox 37, 38, 39
- **System Administration & Diagnostics and Database Programming:** System Administration & Diagnostics is a PC-based application that provides a single interface to quickly access all of the administration and diagnostics tools available on the MiVoice Office 250 system. Database (DB) Programming is installed as part of System Administration & Diagnostics along with other supporting system management tools and utilities.
- **Audio Stream Receive Port:** When upgrading an older system equipped with a PEC-1, ensure that both the system and the network infrastructure are configured to use the appropriate audio stream receive port range (6604–7039) for the PEC-1.
- **Remote upgrades not supported:** Because the chassis LCD panel may prompt you to upgrade the NOR Flash and FPGA by pressing the middle LCD panel button, physical access may be required to upgrade a system.

Upgrade Option One — Replacing An Existing Memory Card

This procedure details replacing an existing v1.2 or later compact flash-type memory card with a new v6.1 or later compact flash-type memory card that ships with the latest core Linux file system and v6.1 or later release software already installed on it.

To replace an existing memory card:

1. Obtain a new v6.0 or later compact flash-type memory card.

2. Perform a Database backup and a Database save using DB Programming. For details, refer to the latest *MiVoice Office 250 Features and Programming Guide*. Also, back up the voice mail data and file-based MOH files (via AWP), if applicable.

   A database backup using DB Programming saves the database to the system. However, you should also perform a *Database Save* to your computer or other accessible location. You can the use this copy to convert, test, and restore the database after the upgrade.

3. Record the system's IP settings for reference.

4. Download the SSL Certificate.

   a. From the **Administrative Web Portal** (AWP) click the **System Information** tab, and then click the **SSL Cert** link.

   b. Click the **Download SSL Certificate** link and click **Download SSL Certificate** button.

   c. Download SSL certificate to your Personal Computer.

   d. Keep downloaded SSL Certificate in safe, protected place.

5. Shut down the system using the LCD panel, and unplug the system power cord.

6. Wearing a properly terminated anti-static wrist strap, remove the existing compact flash-type memory card from the system by pressing the ejector button.

7. Insert the new v6.0 or later compact flash-type memory card.

8. Plug in the system power cord, and power up the system using the LCD panel.

9. If the LCD prompts you to upgrade the NOR Flash and FPGA, press the middle LCD panel button to continue.

   After the upgrade is complete (this process takes several minutes), the system automatically reboots.

10. In its default state, the system attempts to use DHCP for network addressing. If on a network that provides DHCP, press the middle LCD panel button three times to view the current IP address. If on a network that does not provide DHCP, use the LCD panel to update the system IP settings with those recorded in **step 3**.

11. If the system is equipped with an optional Processing Server (PS-1), download the appropriate v5.0 or later PS-1 installer file and follow these steps to upgrade the PS-1 to v6.0 or later:

    For v3.2 or earlier systems:

    a. Record the IP settings for the PS-1 so that they can be manually restored after the upgrade.

    b. Unzip the v6.0 or later PS-1 installer file on to a PC equipped with a CD burner.

    c. Burn the unzipped ISO image file to a CD.

    d. Insert the CD into the PS-1 and reboot the server (it will boot up from the disk).

    e. Follow the instructions displayed on the monitor to upgrade the software.

    f. After the upgrade is complete, reboot the server when prompted, and log on using the "admin" account.

    g. Update the IP settings on the PS-1 with those recorded in step **11a**.

For v4.0 or 5.0 systems only: Upgrade the PS-1 using the standard Administrative Web Portal (AWP) upgrade process. (Or, if desired, use the CD boot upgrade process described above instead of using AWP.)

12. Install the latest System Administration & Diagnostics (which includes v6.0 or later DB Programming) and restore the saved database as outlined on page 319.

> **NOTE** Starting with v4.0, North American compact flash-type cards include both English and Canadian French language prompts. If additional language prompts are required, download and install the desired language packages.

Upgrade Option Two — Upgrading An Existing Memory Card

> **NOTE** Version 5.1 or later requires a minimum one GB compact flash-type memory card.

This procedure details upgrading an existing v3.2 or v4.0 compact flash-type memory card via the Administrative Web Portal (AWP) application. If upgrading a v3.2 or earlier card, the core Linux file system and system software must first be upgraded to v4.0 or v5.0. See the v4.0 or v5.0 documentation for this procedure. The database can convert directly to v6.0. Also note that it can take up to 45 minutes to complete the entire upgrade process.

If, during the upgrade, the LCD panel prompts you to upgrade the NOR Flash and FPGA, press the middle LCD panel button to continue. After the NOR Flash and FPGA upgrade is complete (this process takes several minutes), the system automatically reboots.

To upgrade an existing v4.0 or v5.0 memory card to v6.1 or later:

> **NOTE** When upgrading a v4.0/v5.0 system, the latest core Linux file system was already installed along with v4.0/v5.0, so only the system software is upgraded for v6.1 Release 3.

1. Perform a database backup and a database save using DB Programming. For details, refer to the latest *MiVoice Office 250 Features and Programming Guide*. Also, back up the voice mail data and file-based MOH files (via AWP), if applicable.

   A database backup using DB Programming saves the database to the system. However, you should also perform a database save to your computer or other accessible location. You can then use this copy to convert, test, and restore the database after the upgrade.

2. Record the system's IP settings for reference.
3. Download the SSL Certificate.
   a. From the **Administrative Web Portal** (AWP) click the **System Information** tab, and then click the **SSL Cert** link.
   b. Click the **Download SSL Certificate** link and click **Download SSL Certificate** button.
   c. Download SSL certificate to your Personal Computer.
   d. Keep downloaded SSL Certificate in safe, protected place.
4. From a supported browser, download the v6.0 or later software and associated documentation from Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome) and Mitel eDocs Web site (http://edocs.mitel.com/). Be sure to download the desired NA (North American) or UK v6.0 GA "release" zip file (that includes the latest System Administration & Diagnostics zip file), and (if desired) the PS-1 installer zip file.
5. Unzip the files to a location accessible by your Trivial File Transport Protocol (TFTP) server. Be sure to use a TFTP server that can handle large files. AWP accesses this TFTP server to look for upgrade files.
6. From a browser, enter the IP address for the AWP interface on the MiVoice Office 250 system you want to upgrade.
7. Type the MiVoice Office 250 username and password to log on. The main Web page appears.
8. Click the **System Management** tab, and then click the **System Upgrade** link. A Web page appears.
9. Change the TFTP Address information as follows:
   a. In the first **TFTP Address** box, type the IP address of the TFTP server holding the upgrade files.

      Make sure this IP address points to a separate TFTP server—do not set it to the internal TFTP server on the MiVoice Office 250 system.

   b. In the third box, specify a TFTP server pathname that points to the folder containing the appropriate NA or UK "release" package.
   c. Click **Save Settings**.
   d. Click **View Latest Packages** to view the release packages included in the v6.0 or later release. Confirm that the release file versions listed in the Available column match the ones listed in the release notes included with the software download.

10. Click **Upgrade All**, and then click **OK** at the prompt. After the file transfer is complete, the system reboots automatically and then installs the software release files. (Note that the entire process can take up 20 minutes or longer to complete.) After the release upgrade process is complete and the system reboots again, the LCD panels displays “MiVoice Office 250” (or the programmed hostname for the server).
11. If the system is equipped with an optional Processing Server (PS-1), repeat the upgrade procedure for the PS-1 to upgrade it to v6.0 or later. (Or, if desired, use the CD boot upgrade process described in step 11 on **page 314** instead of using AWP.)
12. After the entire upgrade process is complete and system operation is restored, refresh the AWP browser, and log back on to the system.
13. Click the **System Management** tab, and then click the **Upgrade** link to verify the installed v6.0 or later packages, as indicated in the Version column. An example is shown below.

14. For v5.1 systems only: do either of the following:
    - If you want to make the 86xx IP phone firmware available on the internal TFTP server, point to the proper TFTP path, and do the following:
      a. Click the **System Management** tab, and then click the **Upgrade** link.
      b. Click **View Latest Packages** at the bottom of the page.
      c. Select the check box for the IP phone firmware.
      d. Click **Install** at the bottom of the page.
    - If you want to remove specific language voice prompts you do not need, click the link for the voice prompt file name, and then click **Remove**.
15. Install the latest System Administration & Diagnostics (which includes the latest DB Programming) and restore the saved database as outlined in the following sections.

Upgrade Process Troubleshooting

Should you encounter any problems upgrading software using the AWP application (as described in the preceding section), **Table 56** below summarizes the troubleshooting strategies recommended for resolving discrepancies that may occur when executing the upgrade process.

Table 56: Upgrade Process Troubleshooting Tips

Symptom	Possible Cause	Corrective Action
Upgrade action attempted, but nothing happens.	TFTP server settings configured incorrectly.	Verify TFTP server, port, and path are entered correctly. Ensure files requested are present.
Selecting “View Latest Packages” or “Upgrade All” generates the following error message: Fail: tftp error.	TFTP server settings configured incorrectly.	Verify TFTP server, port, and path are entered correctly.
	TFTP server failure or TFTP server not running.	Verify operation of the TFTP server.
		(Sheet 1 of 2)

Table 56: Upgrade Process Troubleshooting Tips (Continued)

Symptom	Possible Cause	Corrective Action
Selecting “Upgrade All” generates the following error message: Upgrade status file does not exist, unable to get upgrade info.	The browser is unable to find upgrade status information.	Click the back link. If the package summary page shows “To monitor the upgrade, click here” the upgrade is still in progress. Click this link to attempt to retrieve the upgrade status again.
	There was a failure during the upgrade process, and the upgrade has aborted.	Attempt the upgrade once more. If it fails again, contact Mitel Technical Support.
During an upgrade in progress, the web page stops updating.	The browser has lost its connection to the system.	Refresh the browser to attempt to regain the connection.
Download of upgrade component completes, but fails install.	Checksum on one or more of the files will cause install to quit. The system will not be corrupted. Dependency checks may fail. Compact flash-type memory card space requirements could be exhausted.	The possible causes will result in a message at the web browser console (in addition to messages in the system logs). For a checksum failure, check the file on the server, and replace if a checksum failure. Attempt to download again and observe results.
		Log in as Support to check the disk space usage statistics and verify there is sufficient space on the compact flash-type memory card to complete the download.
System is running, but web browser indicates “Unable to connect” or “Cannot display web page”, or System Administration & Diagnostics shows “AWP web page and advanced diagnostics monitoring are not available.”	Web server is not running.	Enable the web server through DB Programming and browse to the network address or connect with System Administration & Diagnostics.
No description or version information available for any of the components.	The package database has been corrupted.	In the Web interface, select View Latest Packages to refresh the browser to retrieve the current package information, then reinstall all packages.
Failure during install process.	There may be a power failure or a technician could inadvertently reset the system during the upgrade process.	If an install was aborted for either of the possible causes cited, the install process can be restarted and the system should restore correctly. However, for NOR flash components or FPGA this action could be fatal, resulting in the system having to be returned to Mitel for repair.
System fails to boot.	Power lost during upgrade	Restore power to the system and reboot it.
A warning appears on the AWP page stating the site is not secure, after system upgrade.	Authority SSL certificate has been lost during system upgrade.	Upload authority SSL certificate.
CAUTION: Do not select Generate Certificate Signing Request (CSR) if you already have the authority SSL certificate, which has been lost during upgrade process. Generating CSR leads to the authority SSL certificate becoming invalid.		
(Sheet 2 of 2)		

Installing System Administration & Diagnostics

Installing the latest System Administration & Diagnostics application automatically installs the latest DB Programming application. DB Programming can then be launched from the System Administration & Diagnostics user interface. Newer versions of DB Programming will later be installed as plug-ins so that the main System Administration & Diagnostics application does not have to be re-installed just to get a newer version of DB Programming.

Instructions for installing and using System Administration & Diagnostics are outlined in the *System Administration & Diagnostics Guide* (part number 550.8125) and the help accessible from the application's user interface.

NOTE To support the Database Reporting feature, the SAP® Crystal Reports® application must be installed during the System Administration & Diagnostics installation.

Restoring the Database

After installing the v6.0 or later system software and the new System Administration & Diagnostics application as described in the preceding sections, use the MiVoice Office 250 Database Utilities to convert, test, and restore the saved database.

NOTE Restoring the database may result in a system reboot.

To restore the database:

1. Launch the System Administration & Diagnostics application.
2. Open, convert, and test a copy of the database that was saved prior to installing the new system software. Use the MiVoice Office 250 Database Utilities to convert the saved database, and then use the DB Test and Repair utility to test it.

   DB Test removes unused items from the database (e.g., unused trunk groups, or extension lists or hunt groups with no assigned extensions). For this reason, you may want to run it in Test mode only, and then view the changes in the log. If the changes will not negatively impact the database, then run DB Test again in Test and Repair mode.

   NOTICE

   You run the risk of running the system on a corrupt database and/or losing e-mail gateway settings if you do not convert and test the database.

3. Restore the saved, converted, and tested database. Also restore any saved voice mail data and file-based MOH files, if necessary.

NOTE When using System Administration & Diagnostics to connect to a system with a restored database, be sure to enter the appropriate system account username and password pair.

4. Upload the license file:
   a. From the DB Programming menu bar, select Operations – Software License Operations – **Upload Software License**.
   b. Enter the path and name of the license file or click **Browse** to search for the file.

c. Click **Start**. The system compares the new software license to the existing license. If you click **Start** before selecting a license (`.isl`), a message appears, informing you that you have not identified a file name.

d. Click **Finish**. If you click **Cancel**, the license is not uploaded.

e. Select Help – About MiVoice Office 250 Software License.

5. Make sure the time zone (under system-wide parameters) is set correctly for the site where this system is located.

6. Make some test calls on the system, and verify basic system functionality.

7. Upload the SSL certificate.

a. From the Administrative Web Portal (AWP) click the **System Information** tab, and click the **SSL Cert** link.

b. Click the **Upload SSL Certificate** and select **SSL certificate from PC** using **Choose File** button.

c. If you previously downloaded SSL certificate from this or different MiVoice Office 250 system, enable flag **Upload a certificate from different MiVo-250 system**. If you use a new signed SSL certificate from CA based on last generated CSR on this MiVoice Office 250 system, disable **Upload a certificate from different MiVo-250 system**.

d. Click **Upload SSL Certificate** button and restart web server.

Expansion Scenarios

The following expansion scenarios are discussed in this section:

- **Upgrading Only Software** below
- Expanding by Adding a PEC-1 on **page 323**
- Expanding by Adding a PS-1 on **page 324**

Upgrading Only Software

Before performing the following upgrade procedures, read “Network Environment Requirements” on **page 1-307** and “Guidelines for Expanding or Upgrading a System” on **page 1-307**. For some of the tasks in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*, as appropriate.

The following process and applicable procedures apply to upgrading only the software version running on the platform.

To upgrade the system version:

1. Obtain applicable licensing from the Applications Management Center (AMC), as necessary.
2. Back up system and voice mail databases. Partial Voice Processor saves are not permitted when upgrading to v2.x or later.
3. Run DB Test & Repair.
4. Use the MiVoice Office 250 Database Utilities tool to convert the database to the correct version and system type.
5. Configure the system to point to the correct USB flash drive (see “New USB Upgrade Option“ on **page 322**) or Trivial File Transfer Protocol (TFTP) server and path, then upgrade the system to the latest release. For specific instructions on upgrading to v5.0 or later , see “Upgrading Existing Systems to v5.x and later Software“ on **page 311**.

   NOTICE

   Possible Data Corruption. Do ***not*** unplug the chassis when the system is in the process of upgrading because critical system data is migrating at this time.

6. Install applicable licensing for the upgraded version.
7. Restore saved databases.
8. Perform a full save of the database and voice processor data after the upgrade is complete.
9. Verify results by performing the following or similar tests:
   - Listen and confirm 2-way audio on IP-to-digital telephone connections and between digital-to-digital telephone connections.
   - Test all trunks, incoming and outgoing.
   - Place incoming and outgoing calls.
   - Exercise the company’s dialing plan.
   - Test hunt group functionality, if applicable.

- Confirm System Manager connections and functionality, if applicable.
- Confirm functionality of all peripherals connected to or dependent on the MiVoice Office 250.
- Confirm that emergency numbers work. However, ***before*** conducting the tests, be sure to coordinate with appropriate local emergency response agencies.

New USB Upgrade Option

To support future upgrades (after v5.0 or later is installed) from a Universal Serial Bus (USB) storage device (such as a USB "flash" drive), a new USB option has been added to the AWP Upgrade page. Previously, only Trivial File Transfer Protocol (TFTP) upgrades were supported. When the USB option is selected, the TFTP form is hidden and the USB form is shown instead (and vice versa). ***Before using the USB option, see "Important Considerations"*** on **page 322**.

The USB upgrade form has a single text box and a "Browse USB…" button. The text box can be used to manually enter a path to the USB storage device where the packages are located. Alternatively, clicking the browse button initiates a search of the selected USB drive for any packages that may exist, and then lists the versions available for installation. Upon selecting one of the versions, the text field is populated with the path.

Clicking the View Package Details buttons reloads the page and shows the current packages installed, available packages, and their versions. If a change is made to the upgrade location, clicking View Package Details or Upgrade Now saves the changes and then performs the selected action. If the upgrade location is changed while viewing package details, a message appears warning that the package details may need to be refreshed.

This USB upgrade option is available with both the Base Server and PS-1, but performing a USB upgrade applies only to the system where the USB drive is physically attached. This means that either two USB drives are needed (one with the Base Server packages and one with the PS-1 packages) or one USB drive with both sets of packages will need to be transferred between systems. Making changes to the upgrade path location on either the Base Server or the PS-1 does not propagate the location to the other server.

Prior to v5.0, changes to the upgrade path location would not be allowed if DB Programming was connected to the system. The ability to change the TFTP server/port/path has been removed from DB Programming in v5.0, so this restriction is no longer be enforced.

Important Considerations

Please note the following when using a USB flash drive for software upgrades:

- **Leave the security key in place:** If the system uses a USB security key instead of a serialized compact flash-type card, leave the USB security key in place and use another available USB port for the USB flash drive.
- **Use only one flash drive at a time:** Besides the USB security key, the MiVoice Office 250 supports only one USB "mount" at a time. So, for example, if you have a USB flash drive connected for database saves or backing up voice mail data, remove it before inserting the USB flash drive containing the upgrade software.
- **Wait ten seconds to swap flash drives:** Because the MiVoice Office 250 supports only one USB mount at a time, you must allow adequate time for the system to unmount/mount devices

when swapping USB flash drives. So, if you need to remove a USB flash drive currently being used for database saves or saving voice mail data, remove the device and ***wait at least ten seconds before inserting the other USB flash drive***. If you do not allow sufficient time for the first device to unmount, the system might be confused as to which USB flash drive is actually plugged into the system.

- **Use multiple folders for multiple applications:** It is possible to use one flash drive for database saves, voice mail data saves, and software upgrades. However, be sure that all of the functions are using different folders on the drive and are not resident at the root level. For example, the folder you select for saving voice mail data automatically gets cleared prior to a save. So, if you have everything at the root level or in the same folder, the software packages used for upgrading will get deleted.
- **Supported flash drives:** The MiVoice Office 250 supports all popular USB flash drive sizes and file systems (e.g., FAT 32 and NTFS). If you come across a USB flash drive that does not appear to be compatible, report the details to Mitel Technical Support.

Expanding by Adding a PEC-1

Before performing the following expansion process and procedures, read “Network Environment Requirements” on **page 1-307** and “Guidelines for Expanding or Upgrading a System” on **page 1-307**.

Parts and Materials

To perform the following expansion process and procedures, you will need:

- Processor Expansion Card (PEC-1), part number 580.2020.
- Applicable licensing.
- Personal computer (PC) with Ethernet/Internet connectivity to the Mitel 5000.
- DB9M to DB9F straight-through cable, part number 813.1683, packed in the Universal RS-232 Kit, part number 828.1282.
- DB9F-to-DB9F null modem adapter with hardware (HW) flow control, part number 804.2605, packed in the Universal RS-232 Kit, part number 828.1282.
- Flash drive(s) for backing up the voice mail database, prompts, and received messages. For a current list of compatible USB flash drives, refer to Knowledge Base article 2468 on the Mitel Knowledge Base Center (accessible through Mtiel Online).
- PC hard drive or other data storage device for backing up the system database so it can be restored after expanding the system.

Processes and Procedures

For some of the tasks in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*.

To expand a system by adding a PEC-1:

1. Perform a *complete* Voice Processor save to one or more thumb drives and a database save to a PC or other adequate data storage device.

2. Run DB Test on the saved database and verify that no problems exist with the database structure.
3. Use the DB Conversion tool to convert the database to the correct version and system type.
4. Run DB Test on the converted database and verify that the database is intact.
5. Configure the system to point to the correct TFTP server and path and perform an UpgradeAll to upgrade the system software.

> **NOTICE**
>
> **Possible Data Corruption**. Do ***not*** unplug the chassis when the system is in the process of upgrading because the Norflash package is migrating at this time.

6. Install a Processor Expansion Card (PEC-1), part number 580.2020. For an HX Controller platform, see "Adding a Processor Expansion Card (PEC-1)" on **page 1-181**. For a CS Controller platform, see "Adding a Processor Expansion Card (PEC-1)" on **page 1-186**.
7. Load the software license for the version installed.
8. Restore the full database and voice processor data to the system.
9. Verify results.
10. Perform a full save of database and voice processor data after the expansion is complete.

Expanding by Adding a PS-1

Before performing the following expansion process and procedures, read "Network Environment Requirements" on **page 1-307** and "Guidelines for Expanding or Upgrading a System" on **page 1-307**.

This section describes the parts and materials and the process and applicable procedures required to add a Processing Server (PS-1) to an existing system.

Parts and Materials

Adding a PS-1 to an existing system requires the following items:

- Processing Server unit, part number 580.5600.
- Applicable licensing.
- Personal computer (PC) with Ethernet/Internet connectivity to the Mitel 5000.
- One of the following input-output setups:
  - A monitor, keyboard, and mouse, *or*
  - A laptop PC, straight-through RS-232 cable terminated with DB-9 connectors, part number 813.1683, and a null modem with hardware flow control, part number 804.2605. The Universal RS-232 Kit, part number 828.1282. For connection details, see Figure 15 on **page 211**.
- Flash drive(s) for backing up the voice mail database, prompts, and received messages. For a current list of compatible USB flash drives, refer to Knowledge Base article 2468 on the Mitel Knowledge Base Center (accessible through Mitel Online).
- Hard drive or other data storage device for backing up the system database so it can be restored after expanding the system.

Processes and Procedures

With the PS-1 chassis installed in accordance with “Installing the Physical Processing Server (PS-1)” on **page 1-204**, you are ready to configure the PS-1 and the Base Server.

To add a PS-1, you must first upgrade the software, then configure the system settings. The conversion process requires that you perform the following procedures.

1. Prepare the Base Server for the conversion and upgrade.
2. Associate the Base Server with the PS-1on the LAN.
3. Configure the PS-1 on the LAN.
4. Configure the Base Server in DB Programming.
5. Reload system and voice mail databases.
6. Verify that external clients point to the PS-1 rather than the Base Server.

To prepare the system for the conversion and upgrade:

1. Back up the system and voice mail databases. Refer to the *MiVoice Office 250 Features and Programming Guide*, for the detailed procedure.
2. To verify the integrity of the saved database, run the DB Test & Repair Utility.
3. Convert the database.
4. Upgrade the system software to the latest version. For the detailed procedure, see “Upgrading Existing Systems to v5.x and later Software“ on **page 311**.
5. While the system is rebooting as a result of the upgrade, remove the USB security key from the Base Server chassis and insert it into a USB 2.0 port on the front or back of the PS-1. See Figure 14 on **page 203** and Figure 6 on **page 221**.

To associate the Base Server and the PS-1 on the LAN:

1. Use the LCD panel menu to navigate to Network\Current\<IP Address> to see if the IP address for the PS-1 has auto-converted to the correct values.
2. If the correct values do not appear, use the LCD panel menu to navigate to Network – Configure – **IP Address**. Enter the Base Server static IP, subnet mask, and gateway IP addresses and *disable* DHCP.
3. Under Main Menu on the LCD panel select System Menu, and then scroll down to Gateway mode. Toggle the Gateway Mode setting from Disabled, to Enabled. Observe the message, Waiting for PS-Server Connection on the LCD panel of the Base Server.
4. Power up the PS-1, if it is not already done. The PS-1 comes with the software pre-installed. After the system has booted up, it performs a self-test. When the self-test completes, the LCD panel on the Base Server displays MiVoice Office 250.

To configure the PS-1 on the LAN:

1. Attach a monitor, keyboard, and mouse or, if using a laptop PC, attach the specified serial cable and null modem to the PS-1. For details see Figure 15 on **page 211**.

2. If attached with a laptop PC, start a hyperterm session and set the connection baud rate at 115200. Then start a PuTTY® session. You will find a copy of PuTTY software on the system software DVD. Read the Readme file and follow the PuTTY installation and user instructions.
3. From the monitor and keyboard or PC attached to the PS-1, log on to the system using the appropriate admin account username and password. The Diagnostics Management Utility screen or window appears.
4. From the DMU main menu select **Configuration**. The Configuration screen appears.

NOTE The PS-1 and Base Server must be connected to the network on the same LAN, VLAN, and subnet. For additional details, see "Network Environment Requirements" on **page 1-307**.

5. From the Configuration screen:

a. Change Static IP Address for the PS-1

b. Change Static Subnet

c. Change Static IP Gateway

d. Disable DHCP

e. Reboot the system

To configure the Base Server in DB Programming:

1. From the PS-1 IP address, start a DB Programming session.
   - If a PS-1 is not connected to a Base Server, the Connection Wizard appears. Follow the instructions in the Connection Wizard to make the Base Server–PS-1 Connection. When you click **Finish**, the Connection Wizard closes the DB Programming session.
   - To continue, restart the DB Programming session.

   For details on using this wizard, refer to the Connection Wizard topic in the DB Programming Help. When you click **Finish**, the Connection Wizard closes the DB Programming session.
2. Upload the appropriate system-specific licensing. From the DB Programming menu bar, select Operations – Software License Operations – **Upload Software License**.

To reload system and voice mail databases:

For the tasks in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*.

1. Restore the full database and voice processor data to the system.
2. Verify results.
3. Perform a full save of database and voice processor data after the expansion is complete.

To verify that external clients point to the PS-1: During the installation and system bring-up process, existing external clients should automatically re-set to point to the PS-1 IP address. To avoid possible interruption of service, confirm satisfactory operation of all IP-connected clients. External clients must point to the PS-1 rather than the Base Server. Clients include, *but may not be limited to*, OAI CT Gateway, OAI applications, Attendant Console, System Manager, DB Programming sessions, external voice mail and Diagnostics Monitor client sessions.

NOTE To reduce the amount of programming changes to IP phones and remote nodes, Mitel recommends that the system IP address should *not* change because IP networking and IP phone configuration values continue to point at this IP address.

PS-1 Licensing with USB Security Key

A USB 2.0 port on the front of the PS-1 accepts the security key. See Figure 14 on **page 203**. The security key holds the MiVoice Office 250 master license information. This master license includes UVM configuration information and, when inserted, disables the UVM license on the Base Server. In this configuration, the Base Server does *not* need a security key.

You must first connect a Mitel USB security key to the PS-1. The security key has a Mitel serial number. The customer must then purchase a software license that corresponds to the Mitel serial number. Once received, the software license can be uploaded to the PS-1 through DB Programming. For details, refer to the *MiVoice Office 250 Features and Programming Guide*.

OPTIONAL SYSTEM EQUIPMENT

This section provides procedures for installing optional equipment on a MiVoice Office 250.

Installing a CT Gateway

System OAI Level 2 applications can be used in conjunction with the CT Gateway. The CT Gateway provides a TCP/IP System OAI interface that supports a multi-node network. Each node in a network functions individually, as far as the System OAI streams are concerned; you cannot program a System OAI serial port on Node 1 that will control devices or receive events for other nodes. If you need to control devices on multiple nodes, you must install a CT Gateway. The CT Gateway receives the OAI streams from multiple nodes and consolidates them into a single stream for a System OAI application. The Gateway sends and receives all of its data using TCP/IP connections. (At this time, each application requires a separate Gateway.)

NOTE Mitel recommends that you upgrade to the latest available CT Gateway version.

For more information about the CT Gateway, see the *CT Gateway Installation Manual*, part number 835.2161.

Installing an External USB Modem

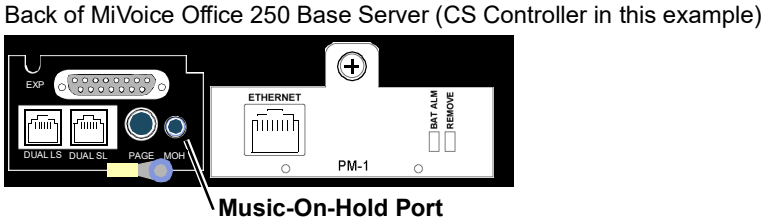
The 5000 Modem Kit (part number 550.3026) has been updated to include a Universal Serial Bus (USB) to Asynchronous (ASYNC) cable (part number 813.1873) that allows you to attach an external modem to the MiVoice Office 250. After all connections are established, you can configure, manage, or debug the system through this modem connection.

For installation and troubleshooting information for the external modem, refer to the *Modem Kit Installation and Troubleshooting for MiVoice Office 250* document (part number 835.1621) included in the Modem Kit.

Installing an External Music Source

External music sources may be connected to the system for Music-On-Hold (MOH). The music jack optional external music sources (radio, tape player, etc.) is on the back of the Mitel 5000 chassis. The music source can be used for music-on-hold and background music.

Figure 5: Music-On-Hold Connection



For the music-on-hold feature, music, silence, ringback, or tick-tones can be assigned on a station-by-station basis. For more information, refer to the *MiVoice Office 250 Features and Programming Guide*.

NOTE There are often broadcast restrictions associated with copyrighted music. Check with the music's original distributor and/or the broadcast source for restrictions and usage concerning background music and music-on-hold. Mitel is not responsible for any illegal or improper use of copyrighted music connected to the system. With European systems, contact PPL (Phonographic Performance Ltd.) for licenses.

Using background music and paging on a hard IP phone, such as the IP PhonePlus, consumes as much bandwidth as a call to the device. If the site has limited bandwidth, ensure that IP phone users do not enable background music and that they limit using the paging functionality.

If using a radio as a music source, place it 5–10 ft (1.5–3 m) away from the chassis to avoid RFI generated by the chassis. If the radio is placed more than 10 ft (3 m) away from the chassis, use twisted-pair cable. For better reception, a radio with an external antenna is recommended.

To install the external music source:

1. Attach a 3.5 mm, two-conductor (mono for CS Controller, stereo or mono for HX Controller), mini-phone plug to one end of a 5–10 ft (1.5–3.0 m) length of shielded speaker cable, or twisted-pair cable, to run from the external music source to the MiVoice Office 250 controller. A stereo plug will not work properly in the MOH jack. If the stereo earphone output of an external music source is used, a stereo-mono adapter should be used between the stereo output jack and the monaural cable.
2. Do one of the following:
   - Connect the other end of the cable to the speaker output terminals of the music source.
   - Attach the necessary connector to the other end of the cable, and plug it into the earphone jack on the music source (if available). If the earphone jack is "padded" so that the optimal volume level cannot be reached, connecting to the speaker output terminals may be more effective.

3. Plug in the AC power cord for the music source. *DO NOT* use the same power outlet used for the Mitel 5000. Turn on the AC power to the music source. Also, Mitel recommends using an UPS.
4. Insert the mini-phone plug into the jack labeled MOH on the back of the MiVoice Office 250 controller. See Figure 5 on **page 329**.
5. From a phone, do one of the following:
   - Select a trunk and dial the number (of one of the other trunks) to call back into the system. Put the call on hold to hear the music.
   - Place an intercom call to another station. Put the call on hold to hear the music.
   - Dial feature code 313. You should hear background music.
6. While listening to Music-On-Hold, adjust the volume on the music source to an acceptable level.

Installing External Paging Equipment

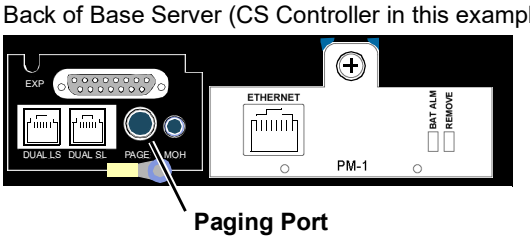
Optional external paging equipment can be connected to the Page port on the back of the Controller. See **Figure 6** for an example of this port location. Use a paging amplifier that *does not* provide talk battery.

NOTICE

Possible Equipment Damage. Using a paging amplifier, with talk battery on a phone, may damage the phone or the module.

Using background music and paging on an IP phone, such as the IP PhonePlus, consumes as much bandwidth as a call to the device. If the site has limited bandwidth, ensure that IP phone users do not enable background music and that they limit using the paging functionality.

Figure 6: Paging Equipment Connection



To install external paging equipment:

1. Cut a length of shielded speaker cable to run from the paging amplifier to the back of the chassis. For an HX Controller chassis, attach a 3.5 mm, two-conductor (mono), mini phone jack. For a CS Controller chassis, attach an RCA-type phono plug to one end of the cable.
2. Connect the other end of the cable to the amplifier high-impedance input according to the manufacturer instructions.
3. Connect the paging speaker(s) to the amplifier using speaker cable.
4. Plug in the amplifier AC power cord. *do not* use the outlet for the Mitel 5000. Also, Mitel recommends using an UPS.
5. Set the amplifier volume control to the lowest setting and turn on the amplifier.
6. Do one of the following:
   - *If installing the paging equipment on a phone speaker or the chassis paging port*:
     a. From a phone location, make a page by lifting the handset, pressing the **PAGE** button (or dialing 7—the default Page feature code), and dialing a zone code that includes the external paging zone or dialing the extension number of the external paging port. The default is 91000.
     b. Adjust the amplifier to the desired level while placing the page.
   - *If installing the paging equipment on a LS circuit:*
     a. Ensure that the LS circuit is not toll restricted.

b. From a phone location, access the LS circuit and adjust the amplifier to the desired level while placing the page.

Installing an External Voice Processing System

For integrated voice processing (such as voice mail, automated attendant, hunt group announcement and overflow, optional Fax-On-Demand, buffered SMDR, Record-A-Call, etc.), an optional external voice processing PC with special applications software can be attached to the system. For additional voice processing information, refer to the voice processing information in the *MiVoice Office 250 Features and Programming Guide*.

NOTICE

As personal computer (PC) technology evolves, Mitel continually enhances the design of its voice processing systems. Depending on the actual version of equipment shipped for your installation, the following instructions may vary. If revised instructions are included with the equipment, see the most recent information.

Introduction

An external voice processing PC can be installed to support increased voice mail capacities and features not available with UVM. External voice processing is provided using NuPoint Unified MessagingTM (UM). For additional information about NuPoint UM systems, including hardware installation, software installation, and upgrading, see the following documentation:

- *MiVoice Office 250 and NuPoint Messenger Integration Guide*, part number 580.8008.

WARNING

Possible Electric Shock. Use the LCD panel to shut down the system, then unplug the MiVoice Office 250 controller power cord before connecting the external voice processing PC. The controller power supply unit operates on 110 VAC and retains a charge after it has been turned OFF. Failure to allow the high voltage to dissipate can cause bodily injury. Like the MiVoice Office 250 itself, the external voice processing PC must have an isolated, dedicated, properly grounded AC circuit for proper operation and it should be plugged into an UPS.

IMPORTANT As with the MiVoice Office 250, the external voice processing system must be connected to an isolated, dedicated, properly grounded AC circuit and UPS for proper operation. The UPS helps to avoid hard drive corruption in the event of a power loss.

Attaching an External Voice Processing System

This section provides information about attaching an external voice processing system to the MiVoice Office 250 controller. For information about installing and setting up the available external voice processing systems, see the documentation provided with those units.

For an illustration of how an external voice processing PC attaches to the MiVoice Office 250 controller, see Figure 3 on **page 336**. For DB Programming tasks in this procedure, refer to the *MiVoice Office 250 Features and Programming Guide*.

To attach an external voice processing system:

1. Back up Unified Voice Messaging (UVM).
2. Disable UVM.

3. Install the external voice processing PC in the rack under the MiVoice Office 250 controller. If the voice processing PC cannot be placed in the rack, place it in the appropriate proximity to the rack. The distance is limited by the connection cable provided. As necessary, follow the instructions included with the voice processing PC you purchased.
4. Attach the Ethernet cables from both chassis to the LAN, as shown in Figure 3 on **page 336**. Use a switch if necessary; do not use a hub.
5. Attach the cable, part number 813.1806, from the external voice processing PC to the MiVoice Office 250 controller.

NOTICE

External Voice Mail Inoperability. If you have an existing external voice processing system (NuPoint orUM), you cannot use the existing cable to connect the external voice processing system to the MiVoice Office 250. You must use cable part number 813.1806.

6. Program the external voice processing system to connect to the MiVoice Office 250 controller. For details, see the external voice processing system manual.
7. Use the Administrative Web Porta (AWP) pages to verify that Unified Voice Messaging (UVM) is disabled.

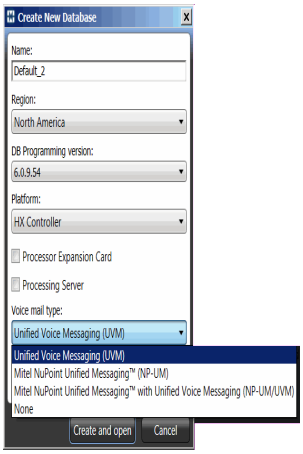
Review and Update for Enabling a Voice Processor

The MiVoice Office 250 supports Unified Voice Messaging (UVM) and NuPoint Unified Messaging (UM). And, for proper configuration, DB Programming requires that the voice processor type is identified. After being connected, the system senses the voice processing type. Connection with a NuPoint UM is over IP.

As with previous versions, a voice processor connected to a MiVoice Office 250 auto-enables when the system boots up. To avoid the risk of overwriting an established database with default values, Mitel recommends connecting the voice processor first, then connecting with a online mode session of DB Programming, and then saving.

To save a voice processor database in online mode:

1. Connect the PS-1 or NuPoint UM equipment to the system in accordance with Mitel installation instructions. Refer to the applicable installation manual.
2. In online mode, launch the System Administration & Diagnostics application and start a DB Programming session.
3. Select the voice mail system to use.

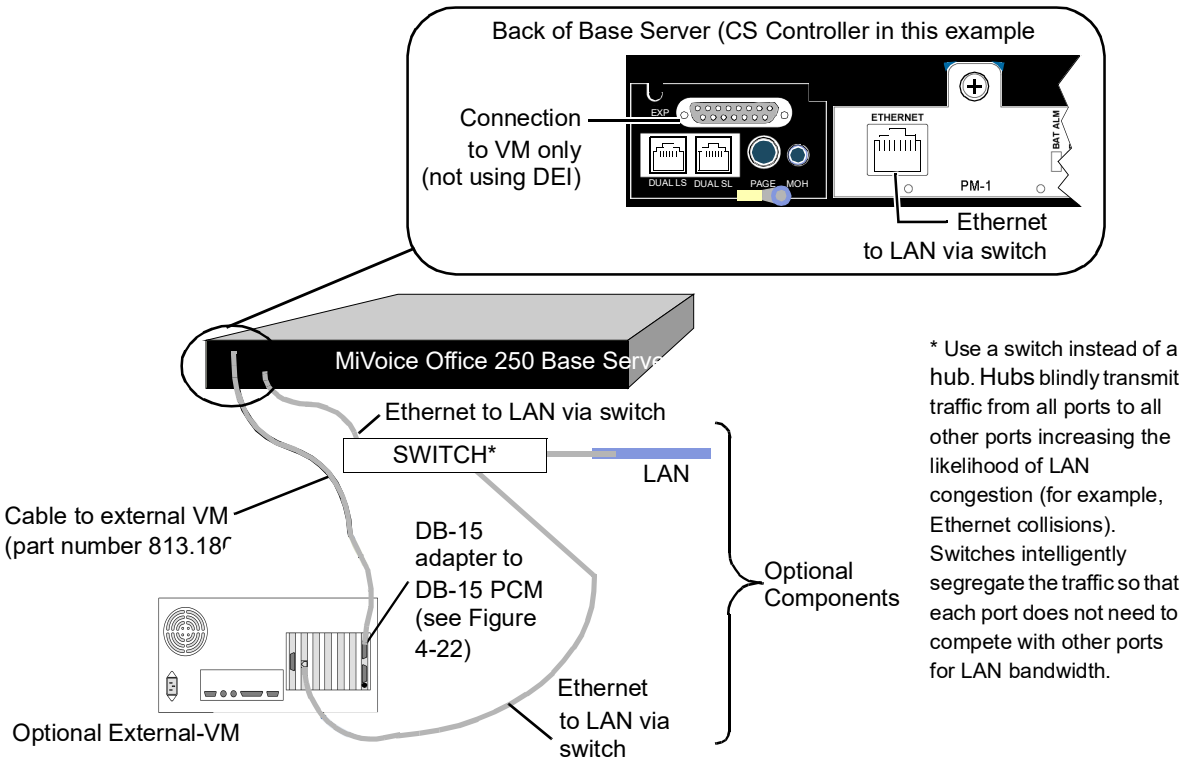


4. From the DB Programming menu, select **Operations** – **Database Operations** – **Backup Operations** – **Save Backup**. Identify the location where you want to store the system database, then select **Save Voice Data to:** box and identify the location where you want to save the voice processor database.

5. Click **Start**. System and voice processor databases are saved to the locations indicated.

Figure 3 shows how an external voice processing system connects to the CS controller. An 8.5 ft (2.5 m) cable is available (part number 813.1806). Or, if necessary, contact the Mitel DataNet CommSource division for information on purchasing longer cable lengths.

Figure 3: Example External Voice Mail Connection to the MiVoice Office 250



* Use a switch instead of a hub. Hubs blindly transmit traffic from all ports to all other ports increasing the likelihood of LAN congestion (for example, Ethernet collisions). Switches intelligently segregate the traffic so that each port does not need to compete with other ports for LAN bandwidth.

Installing SIP Gateways

Support is provided in the system for local Public Switched Telephone Network (PSTN) connectivity using the IP resources in the MiVoice Office 250 controller and a third-party Session Initiated Protocol (SIP) gateway. The IP Resource application expands the support of 25 IP phones/Single Port LSAs/SoftPhones to include Loop Start Adapters (Single Port LSAs). Support for IP-based local loop requires no software feature units.

Up to 25 IP resource ports can be programmed to be loop start adapter ports (SIP/MGCP gateway trunks). The IP Resource application statically maps IP resources on a one-to-one, non-blocking basis for each loop start port within the SIP/MGCP device.

For a current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). You must first log on to Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome) in order to access the KB using this link.

WARNING

Possible Delay in Local Emergency Response to Remote Sites.

IP and SIP phone users should be alerted to the following hazardous situations:

- If an Emergency Call phone number is dialed from an IP or SIP phone located at a remote site that is ***not*** equipped with a correctly configured gateway, the call will be placed from the location where system chassis is installed rather than from the location where the emergency call is made.

  In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of SIP/MGCP gateway trunk(s) for dial tone.
- If uninterruptible power supply (UPS) protection has ***not*** been installed as part of the MiVoice Office 250, IP and SIP phones will ***not*** operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built-in to the system chassis. If a phone connected to a power failure bypass circuit is *not* available, users should make emergency calls from a local phone ***not*** connected to the system. For details about the Power Failure Bypass feature, refer to the *MiVoice Office 250 Features and Programming Guide*.

When configured with a SIP gateway, you can program the system so that when an Emergency Number is dialed from an IP device, the calling name and number associated with the IP device, not the system, is sent to the emergency call operators.

With regard to the *Service Type* feature, call processing supports Caller ID on the Single Port LSA. The Single Port LSA has built-in Caller ID receivers. It does not use shared Caller ID resources within the system. The resource manager within Call Processing is modified to have Caller ID receivers as an attribute for this device type and to use the local Caller ID receiver. If the service type is set to Caller ID, Caller ID messages are passed up through the IP resource application to Call Processing. By default, it is set to None.

IMPORTANT	Your use of Mitel products and/or services, including but not limited to certain caller ID features, may be subject to US, UK, and Canadian regulations and laws. For example, transmitting misleading or inaccurate caller ID information with the intent to defraud or deceive may be a violation of the Truth in Caller ID Act of 2010. Mitel does not promote use of its products and/or services in such a manner and any such use by you which knowingly contravenes local or national governments laws is at your own risk. Mitel strictly disclaims any liability which may be associated with your wrongful use.

NOTICE

It is the responsibility of the organization and person(s) performing the installation and maintenance of MiVoice Office 250 systems to know and comply with all regulations required for ensuring Emergency Outgoing Access at the location of both the main system and any remote communication phones. Remote IP and SIP phones may require gateway access to nearby emergency responders.

Emergency Call phone numbers include:

- 911, the default for Mitel systems located in the United States (USA).
- 999, the default for Mitel systems located in the European market and used primarily in the United Kingdom (UK).
- If applicable, 112, an emergency number used widely in Europe outside of the UK.
- Any emergency number, such as for a police or fire station, that is appropriate for the location of the main system and/or remote phones.

Caller ID receivers are not available on all SIP gateways. Consult the device documentation for information on Caller ID support. Refer to the *MiVoice Office 250 Features and Programming Guide*, for additional details. Figures **4** and **5** illustrate a remote site without or with an SIP gateway.

Figure 4: Example of a Remote Site without a SIP Gateway Application

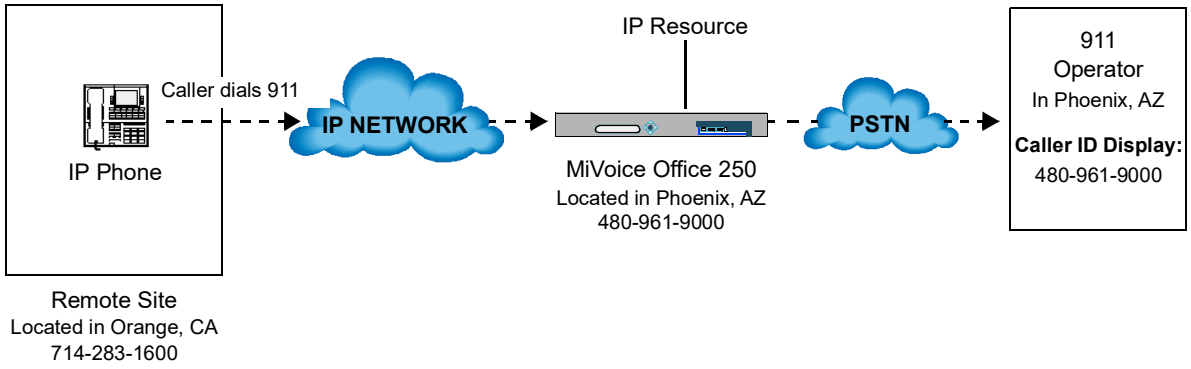
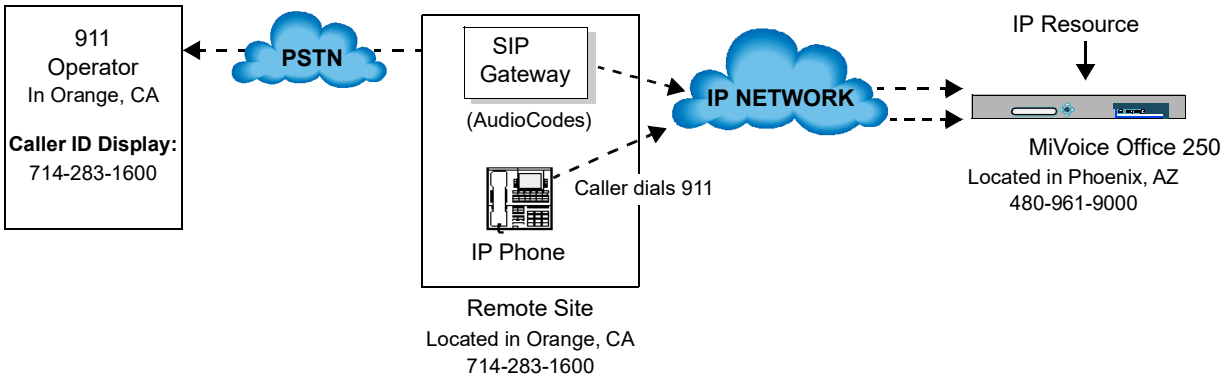


Figure 5: Example of a Remote Site with a SIP Gateway Application



IMPORTANT The SIP gateway is intended to connect to the PSTN. Connecting the SIP gateway to single line ports on other systems is ***not*** supported.

NOTICE

Positive Disconnect. Signaling a call termination from the CO over an analog loop start line differs from CO to CO. Because the SIP/MGCP gateways detect this call event differently, you must pay particular attention to how the CO and the SIP/MGCP gateway detect far-end call termination. If the SIP/MGCP gateway you are installing relies on the CO to detect dial tone to signal a call termination, then a user could possibly hookflash and place a second call before the system realizes that a new call is being made. If this occurs, the appropriate toll restrictions are not applied to the new call. If the CO detects dial tone to indicate a terminated call, Mitel recommends that the CO line used with the SIP/MGCP gateway be restricted to local calls.

Also, if the CO caller hangs up before the user, the user may still be connected to the CO until the user either hangs up or the SIP/MGCP gateway detects dial tone. The user may eventually hear reorder tone or a message that says, “If you would like to make a call...” until they hang up.

The system supports SIP (Session Initiated Protocol) trunks to reach the CO. SIP trunks allow the system to communicate with the CO via SIP-enabled gateways. As the SIP protocol becomes more and more popular, it is important to be able to communicate to SIP gateways in the IP-centric world.

SIP gateways are available through the Mitel DataNet CommSource division by these companies:

- Quintum®
- AudioCodes™

For a current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm). You must first log on to Mitel Online (http://portal.mitel.com/wps/myportal/MOLHome) in order to access the KB using this link.

These gateways replace the AudioCodes MP-104 SIP/MGCP Gateway that has gone end-of-life in October 2006. Both gateways support analog Foreign eXchange Office (FXO) ports. The setup procedure is different for each device; the MP-114 gateway uses a Web interface called the AudioCodes Web administrator, while the AFT 400 gateway uses a Configuration Manager program and a Command Line Interface (CLI). Existing systems must upgrade to v2.4 or later to support the

gateways. For more information, refer to the *Inter-Tel Axxess/5000 Quintum SIP Gateway Installation Manual* (document part number 835.3123) or the *Inter-Tel Axxess/5000 AudioCodes MP-114 SIP Gateway Installation Manual* (document part number 835.3202).

SIP trunks support the following functionality:

- They are transparent to the system user because SIP trunks act like any other CO trunk in the system.
- They support transferring trunks, putting trunks on hold, and connecting trunks to conferences similar to other CO trunks in the system.
- They support making and receiving calls by any phone.
- They support peer-to-peer audio by IP phones.
- They reside in CO trunk groups like other trunks so that SIP trunk calls can be routed using Automatic Route Selection (ARS).
- They support Emergency Number calls, such as 911 in the United States or 999 in Europe, the same as SIP/MGCP trunks.

Installing a PKM 12 or PKM 48

The 12-button and 48-button Programmable Key Module (PKM) are digital devices providing 12, 48, or 96 additional personal buttons for the 5224/5324 IP phones and the newer 8528/8568 digital telephones. And, as of the v5.0 software release, the 5330 and 5340 IP phones now also support the 12-button and 48-button PKMs.

Each personal button may be programmed as a feature button, a Direct Station Select (DSS) button, a Speed Dial button, or for other uses, and each has a Line Status Indicator that behaves the same as the indicators on the attached phone. Up to two PKM 48s may be attached to the phone for a total of 96 additional buttons.

NOTE PKM 12 and PKM 48 are not compatible. You cannot connect both a PKM 12 and a PKM 48 at the same time.

PKM Interface Module (IM)

The PKM IM provides connectivity between the phone and the PKM 12 or PKM 48. The PKM IM is installed separately at the base of the phone. You can connect a phone to up to two PKMs using the PKM IM. The universal 24 VDC power adapter powers the PKM IM and the PKM 12 or PKM 48. The DC adapter is always required to power the PKM IM when used on an 8528/8568 phone.

You need the following items to install a PKM:

- One 5224/5324/5330/5340/8528/8568 phone
- One PKM 12 or PKM 48 (or two PKM 48s)
- One modular cable (or two modular cables for two PKM 48s)
- One PKM IM v2 (v1 is not compatible)
- One Universal 24 VDC power adapter
- One power cord
- Phillips screwdriver

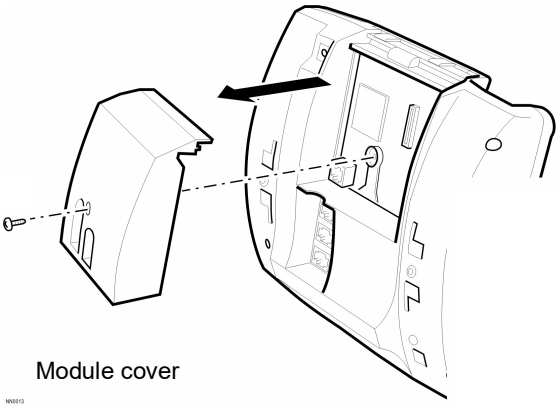
See Table 66, “DSS/BLF Units, PKMs, Data Port Modules, and Transformers,” on **page 375** for the part numbers that are associated with PKMs.

To install the PKM IM:

NOTICE

Possible Device Damage. Do not plug a power supply into a PKM IM installed in the phone. Handle the PKM IM by the cover. Do not touch components or circuitry.

1. Unplug the phone from the system.
2. With a Phillips screwdriver, remove the screw from the module cover from the under side of the phone. Store the cover and screw.

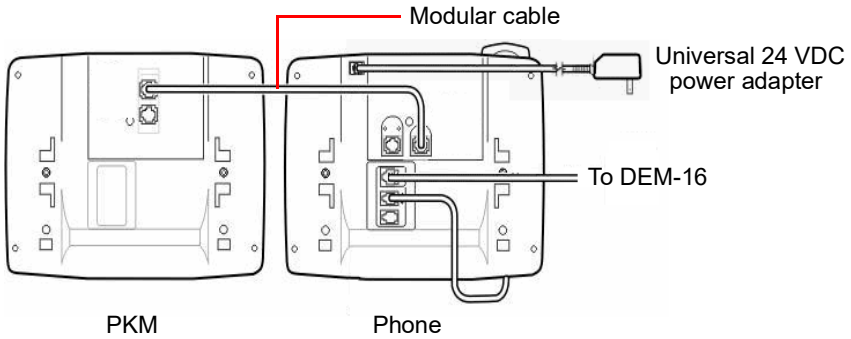


3. Place the PKM IM and apply equal pressure to all sides of the cover until the PKM IM clicks into place.

4. Fasten the PKM IM cover with the supplied screw.
5. Reconnect the phone to the system, and then plug the universal 24 VDC power adapter into the phone's port.

To attach a PKM to the phone:

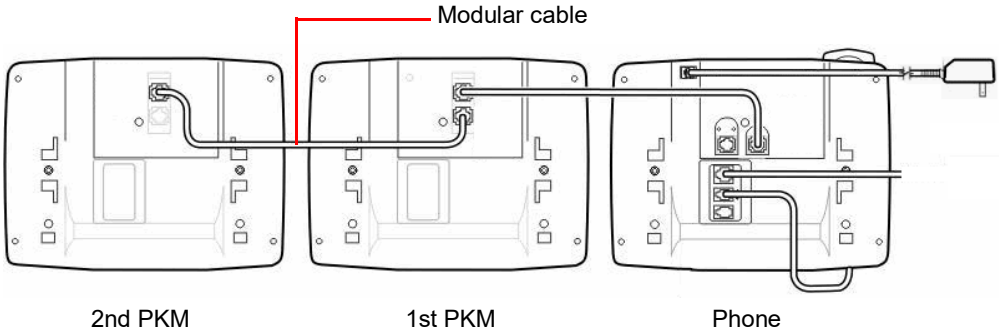
1. Ensure that a PKM IM is installed in the phone as described in the previous section.
2. Plug one end of the modular cable into the jack located on the underside of the PKM.



3. Plug the other end of the modular cable into the jack on the phone's PKM IM. The LEDs temporarily flash and then go out. Continually flashing LEDs indicate incorrect installation or programming.
4. Route and bundle excess cable, and then tuck it underneath the phone stand.
5. Ensure that the PKM is configured in DB Programming as described in the *MiVoice Office 250 Features and Programming Guide*.
6. Program the buttons as described in the appropriate phone user guide.

To attach a second PKM to the phone:

1. Ensure that the phone and first PKM are configured and powered as described in the preceding procedure.
2. Plug one end of the modular cable into the jack on the underside of the first PKM.



3. Plug the other end of the modular cable into the jack on the underside of the second PKM. The LEDs temporarily flash, and then go out. Continually flashing LEDs indicate incorrect installation or programming.
4. Bundle cables, and then program the second PKM as desired.

Installing a PKM 16 (Mini-DSS Unit)

The Programmable Key Module 16 (previously known as the Mitel 8416 Mini-DSS Unit) provides 16 additional user-programmable buttons and connects directly to the Mitel 8520, 8560, and 8660 phones. PKM 16 units are physically attached to a special cable pin connector. Because the PKM 16 draws power from the phone, external AC transformers are not required. PKM 16 units are optional equipment.

Unpack and inspect the PKM 16. Install each PKM 16 as outlined below. For diagrams of the installation, see Figure 6 on **page 345** and Figure 7 on **page 345**.

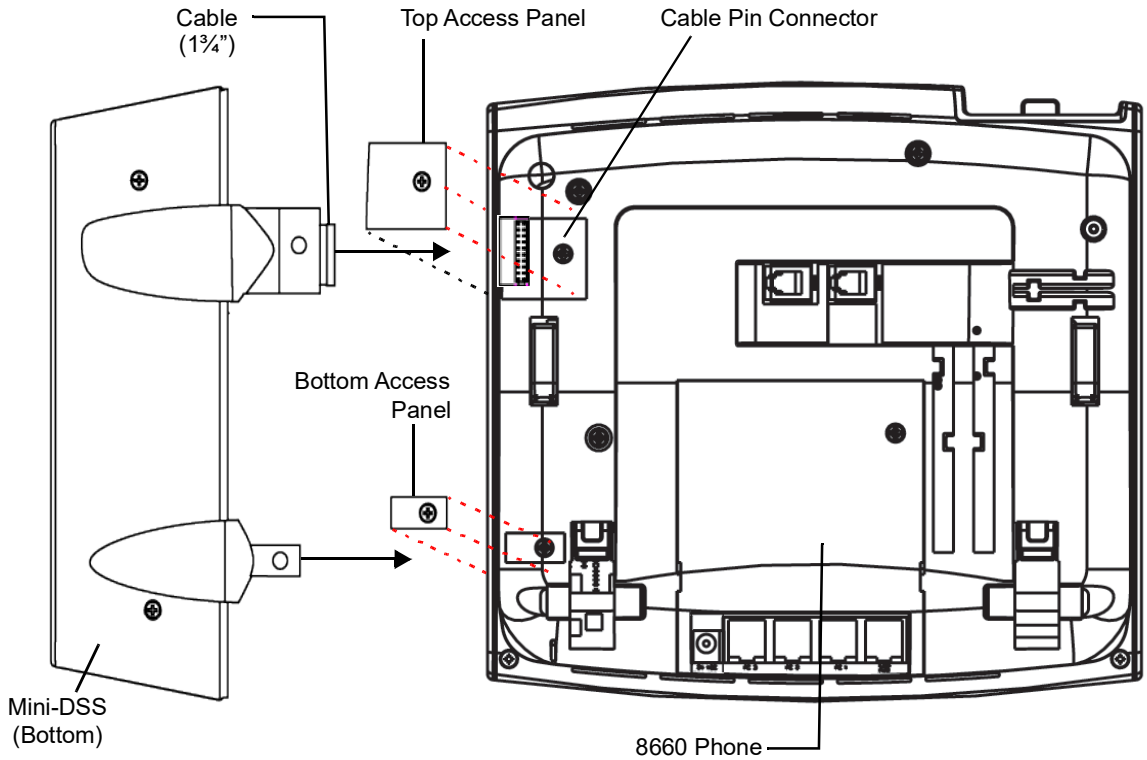
To install a PKM 16:

1. Lift and remove the original faceplate using the notch on the faceplate edge of the phone. The original faceplate can be returned to Mitel for recycling.
2. Remove the original phone facemats and set them aside for later use.
3. Turn the phone over so that the PKM 16 access panels located on the bottom of the phone are accessible. The recess beneath the top access panel houses the pin connector for the PKM 16 cable.
4. Remove the screws from the access panels and set them aside for later use.
5. Remove the access panels from the phone. The phone access panels can be returned to Mitel for recycling.
6. Turn the PKM 16 over and pull a section of ribbon cable measuring 1¾ in. (4.5 cm) from the unit.

NOTE Avoid removing more cable than the recommended 1¾ in. (4.5 cm). The excess cable may be difficult to return to inside the PKM 16.

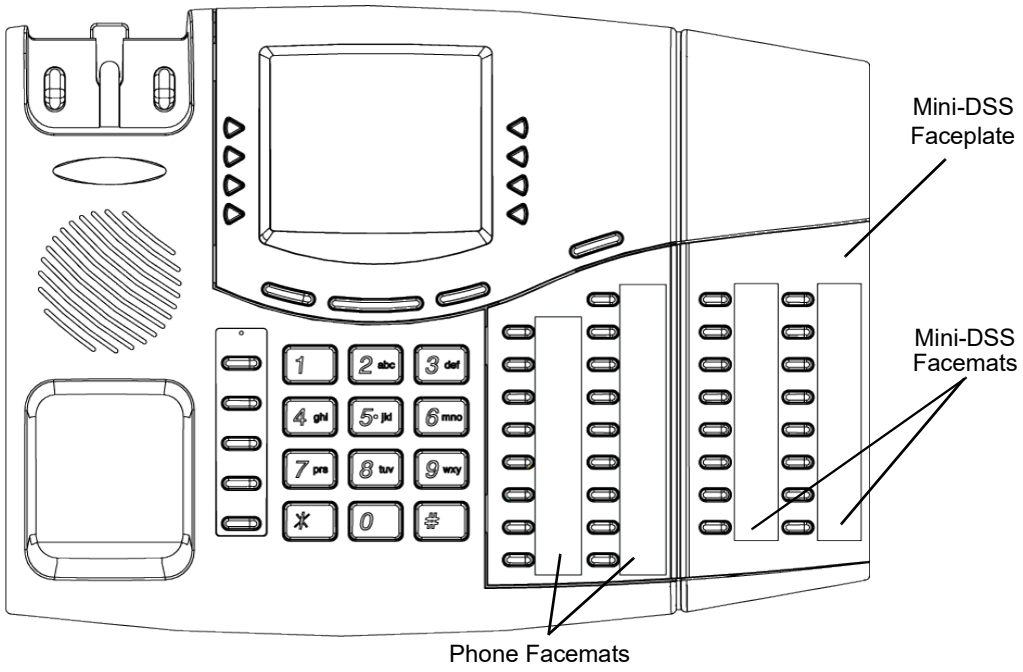
7. Plug the cable from the PKM 16 into the pin connector on the phone.
8. Guide the “arms” of the PKM 16 into the access panel slots.
9. Secure the PKM 16 to the phone by replacing and tightening the screws removed from the access panels in **step 5**.
10. Turn the phone, and attached PKM 16, over. Replace the original facemat on the phone and place the PKM 16 facemat on the PKM 16.
11. Snap the PKM 16 faceplate into place covering the facemats.

Figure 6: PKM 16 (Mini-DSS Installation)



For illustration clarity, the phone base is not shown.

Figure 7: Mitel 8660 with Attached PKM 16 (Mini-DSS)



Installing Other PKMs (DSS/BLF Units)

Three models of Programmable Key Modules (PKMs) (previously known as Direct Station Select [DSS]/Busy Lamp Field [BLF] units) can be installed on digital telephones, depending on the model of phone. Each suitable digital telephone supports up to four PKM (DSS/BLF) units, and each MiVoice Office 250 supports a total of 20 PKM (DSS/BLF) units.

As described in the following sections, to install a PKM (DSS/BLF) unit on a digital telephone, carry out the following process and associated procedures:

1. Gather installation equipment and materials.
2. Install Personal Computer Data Port Module (PCDPM) and Modem Data Port Module (MDPM) devices in digital telephones, as required.
3. Connect PCDPM cables.
4. Install and test PKM (DSS/BLF) unit(s).

Gathering Equipment and Materials

Keep the following information in mind when gathering equipment and materials for installing PKM (DSS/BLF) units.

- PCDPMs are required on digital telephones in order to use the Off-Hook Voice Announce (OHVA) feature or to install PKM (DSS/BLF) units.

  IMPORTANT Serial data communication through the RS-232 serial port interface on the PCDPM is *not* supported on the MiVoice Office 250.

- MDPMs are required on digital telephones in order to connect single line phones.
- Optional wall-mount kits for the Baser Server/DEI chassis. One wall-mount kit allows you to mount the chassis flat against the MDF and the other kit from the Mitel DataNet CommSource division provides a rack mount unit that can be wall-mounted on the MDF. For detailed information about wall-mounting the chassis, see “Installing the CS Controller Chassis” on **page 1-144**.

Installing a PCDPM

The following areas must be addressed to properly install and configure the PC Data Port Module (PCDPM):

- Hardware Installation (discussed on the pages that follow)
  - Executive Display or Standard Display Phone (**page 349**)
  - Professional Display or Associate Display Phone (**page 350**)
  - Mitel 8520 or 8560 Phone (**page 352**)
- System DB Programming (refer to the *MiVoice Office 250 Features and Programming Guide*)

For Off-Hook Voice Announce (OHVA) receive capability, and to allow a PKM (DSS/BLF Unit) and/or Modem Data Port Module to be installed, any Standard, Associate, Executive, or Professional Display phone or Mitel 8520 or 8560 phone may be equipped with a PC Data Port Module (PCDPM). For the Executive and Standard Display phones, PCDPM part number 550.3018 [PB2585 in Europe]

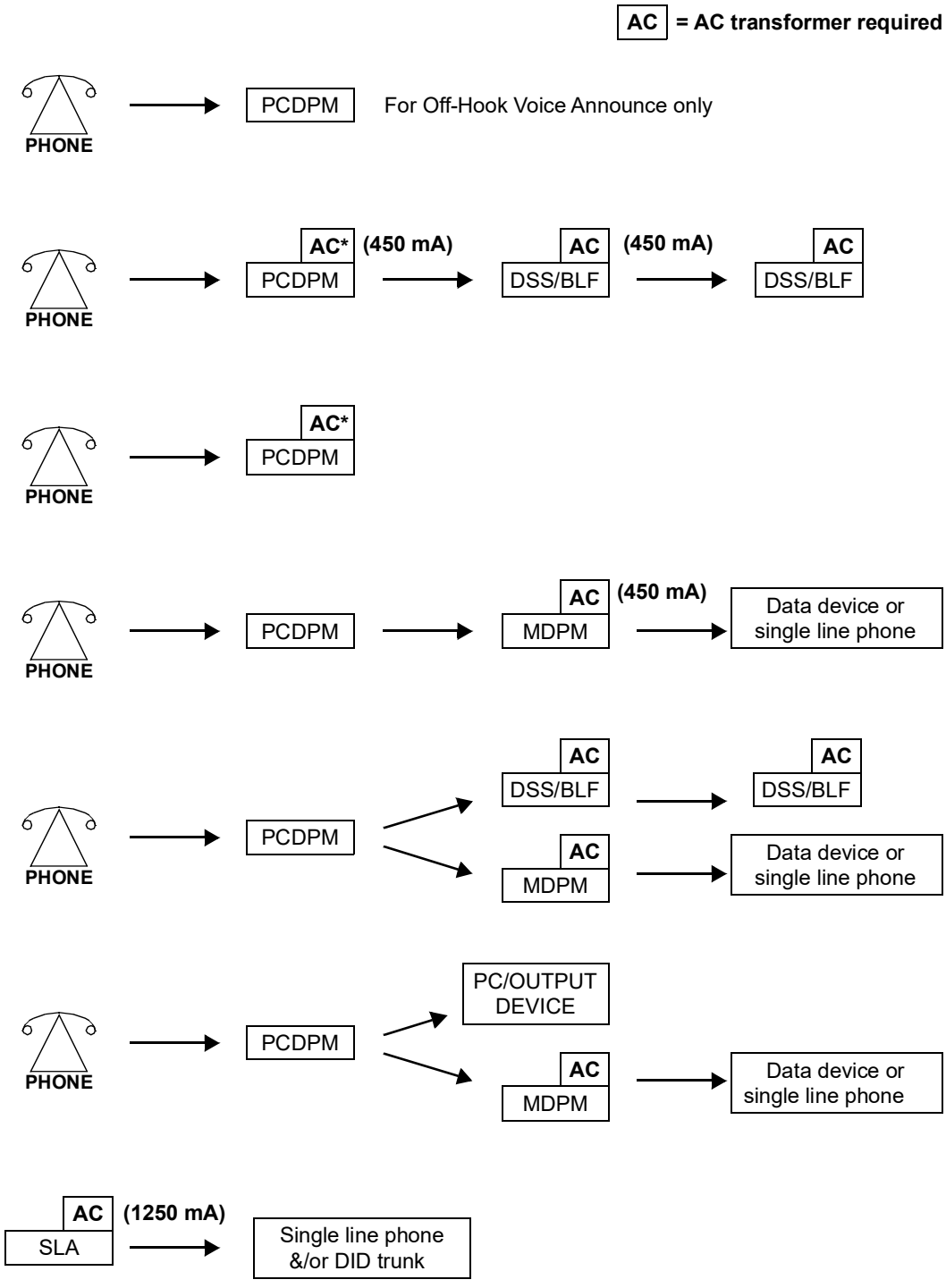
or 550.3014 [PB2548-A] is required. For the Professional and Associate display phones, PCDPM part number 550.3018 is required. Mitel 8520 and 8560 phones require PCDPM part number 550.8400.

The original model PCDPM, part number 550.3014 [PB2548-A in Europe], for the Executive Display and Standard Display phones has been replaced with part number 550.3018 [PB2585]. The newer PCDPM, part number 550.3018, and the Mitel 8520/8560 PCDPM, part number 550.8400, do not require an external AC transformer. For details, see Figure 8 on **page 348**. Refer to the installation instructions included with the unit. Procedures for installing PCDPMs begin on **page 349**.

PCDPMs can be used for the following purposes:

- Receive capability for OHVA
- Connect Modem Data Port Module (MDPM)

Figure 8: AC Transformer Requirements for PCDPMs



* The AC transformer for the PCDPM is only required if using part number 550.3014 [PB2548-A].

IMPORTANT	PKMs (DSS/BLF Units), MDPMs, and SLAs always require separate AC transformers. Older PCDPMs, part number 550.3014 [PB2548-A], require AC transformers when they are connected to PKMs (DSS/BLF Units). PCDPMs do not require AC transformers when they are used for OHVA only or have attached MDPMs.

In Executive Display or Standard Display Phones

For installation diagrams, see Figure 9 on **page 353** and Figure 10 on **page 354**.

To install the PCDPM for an Executive or Standard Display Phone:

1. Unplug the line cord and the handset from the modular jack on the back of the phone.
2. Remove the phone baseplate.
3. Using a pair of cutting pliers, remove the plastic knockout on the back of the phone that covers the PCDPM connector (**J3**) on the control board. See Figure 9 on **page 353**.

IMPORTANT	Be careful *not* to let the plastic knockout fall inside the phone. If the plastic knockout does get inside the phone, do ***not*** remove it. If the phone will be wall-mounted, use the knockout in the bottom half of the phone instead. See Figure 9 on **page 353**.

   Some of the newer-style digital telephones equipped with both red and green LEDs will not have a plastic knockout in the upper portion of the back of the phone to allow easy connection to the control board. Instead, you must open the back of the phone (secured by four screws) and route the cable through the plastic knockout in the lower portion of the back housing, just as you would if you were wall-mounting the phone.
4. Align the PCDPM over the four posts protruding from the inside of the baseplate and insert the screws (do *not* over-tighten).
5. *If not already installed*, attach one end of the PCDPM-to-keyset interface cable to the connector on the back of the PCDPM.

   On older-style digital telephones equipped with red LEDs only, use the appropriate end of the new PCDPM to old keyset interface cable, part number 813.1633. The connector is keyed so that it can be installed in only one way.

   On one of the newer-style digital telephones equipped with both red and green LEDs, use the appropriate end of the new PCDPM to new keyset interface cable, part number 813.1632. Pin 1 of the cable (indicated by the stripe on the edge of the cable) must correspond to pin 1 on the KEYSET connector (indicated by the silk-screen label on the PCDPM). The connector is keyed so that it can be installed in only one way.
6. Plug the free end of the PCDPM-to-keyset interface cable into connector **J3** on the control board—where the plastic knockout was previously removed. Use the eraser end of a pencil or other suitable rod-like instrument to press each end of the cable connector into connector J3. Apply pressure, one end at a time, until the connector clicks securely into place.

IMPORTANT	Pin 1 of the cable (indicated by the arrow on the connector) must correspond to pin 1 on the RS-232 connector (indicated by the silkscreen on the PCDPM). Also, the PCDPM-to-DSS interface cable (part number 813.1595; included with the unit) is not used with the newer model PCDPM; use the DSS-to-DSS interface cable instead.

7. Plug the appropriate end of the PCDPM-to-MDPM interface cable, part number 813.1566 included with the MDPM, into the MDPM connector (**J1**) on the PCDPM. The other end of the cable will be attached to the MDPM after it is installed.

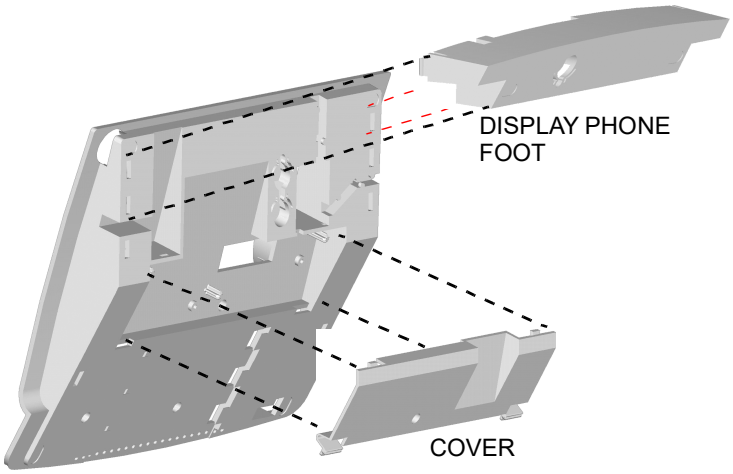
   Pin 1 of the cable, indicated by the stripe on the edge of the cable, must correspond to pin 1 on the MDPM connector, indicated by the silk-screened notch on the PCDPM. Some older PCDPM-to-MDPM cables may have the stripe on the wrong edge. If the MDPM doesn't respond properly, try reversing the connector on the PCDPM.

8. Carefully thread the cables connected to the PCDPM through the slots on the access panel. Re-attach the baseplate and reconnect the line cord.

In Professional Display or Associate Display Phones

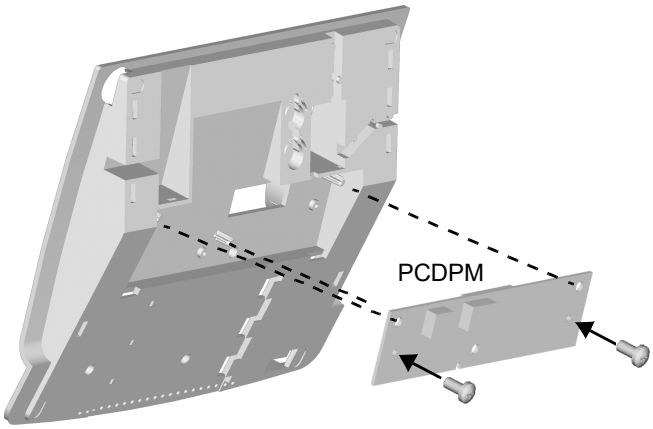
To install a PCDPM in a Professional or Associate Display Phones:

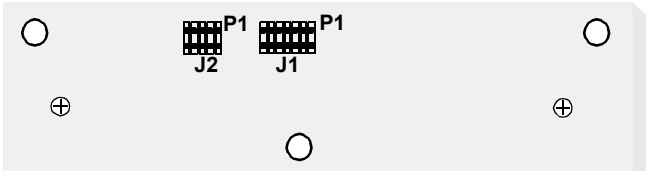
1. Press up on the tabs and lift off the cover. Remove the cover and the display phone foot in the back, as shown below.



2. Align the PC Data Port Module over the connector inside of the display phone and the three pins on the outside. Gently press the module into place until the connector is securely positioned into place, as shown below.

3. Attach the module to the display phone using the two mounting screws.



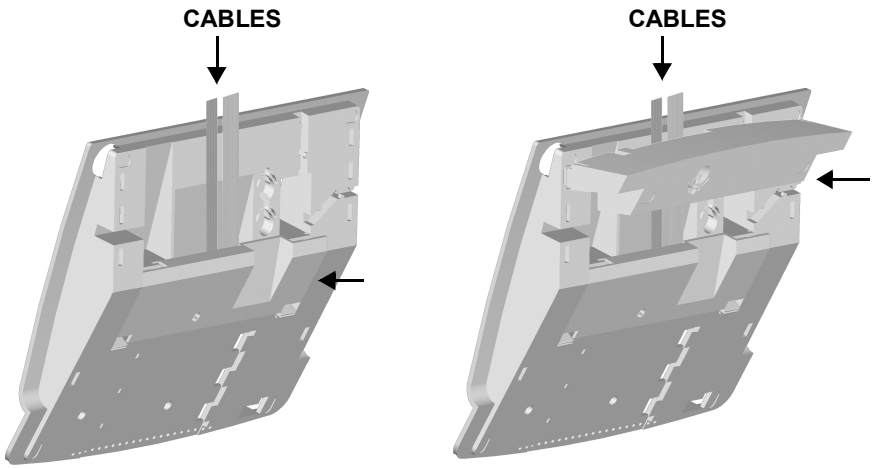


4. Plug the appropriate end of the PCDPM-to-MDPM interface cable, part number 813.1566, into the MDPM connector (**J1**), shown above, on the module. The other end of the cable will later be attached to the MDPM after it is installed.

   Pin 1 of the cable, indicated by the stripe on the edge of the cable, must correspond to pin 1 on the MDPM connector, indicated by the notch silk-screened on the module.

5. Carefully thread the cables connected to the PCDPM through the slots on the access panel, and re-attach the access panel to the phone. Make sure the cables exit through the slot between the cover and the bottom of the display phone, as shown below on the left.

6. Place the cables flat against the phone and carefully place the foot over the cables, as shown below on the right. For sample display phone PCDPM cable connections, see Figure 11 on **page 354**.



In Mitel 8520 or Mitel 8560 Phones

For installation diagrams, see Figure 12 on **page 355** and Figure 13 on **page 356**.

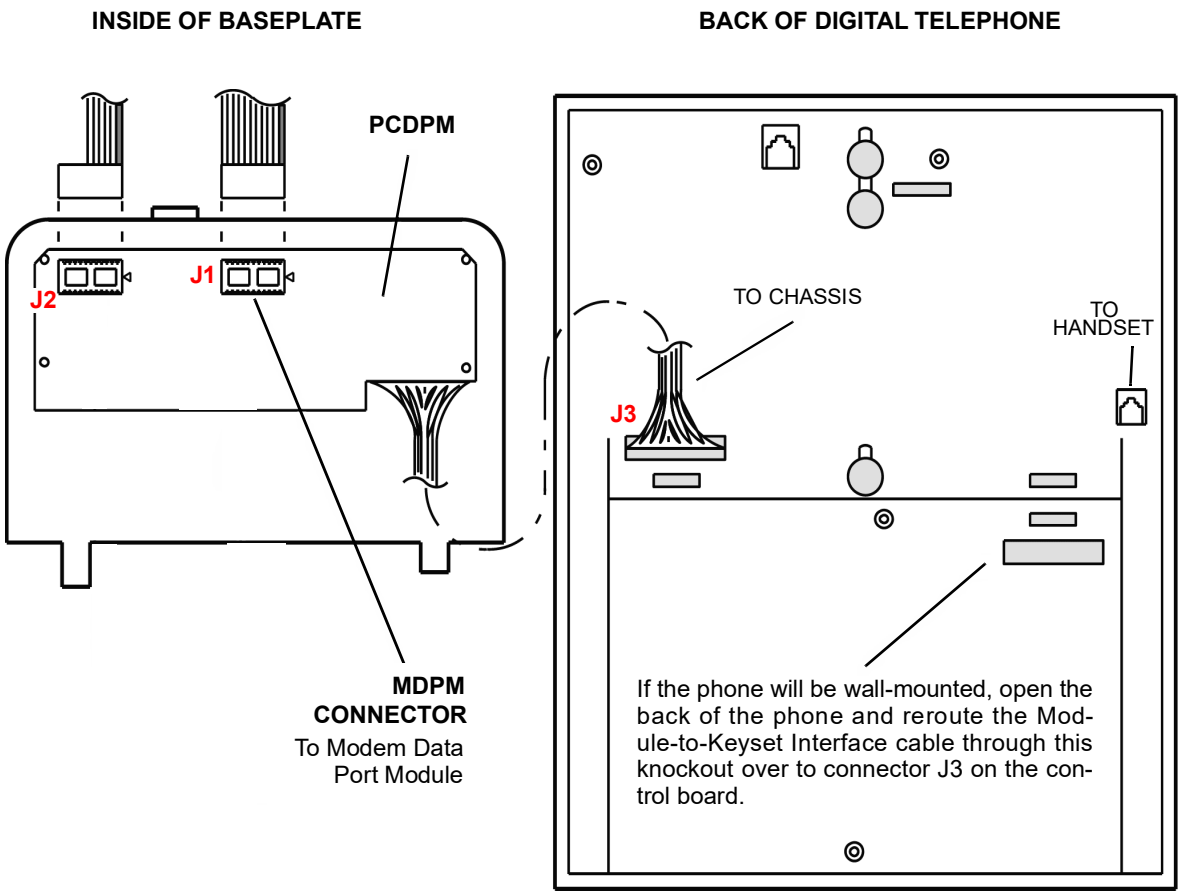
To install the PCDPM in a Mitel 8520 or 8560 phone:

1. Unplug the line cord and headset from the modular jacks on the back of the phone.
2. Open the phone base to a fully extended position.
3. Gently slide the access panel off the phone to expose the PCDPM pin connector (**J10**) on the phone control board.
4. Align the PCDPM **J3** connector with the board **J10** connector and plug the PCDPM into place on the phone board (see Figure 12 on **page 355**).

   If a Modem Data Port Module will be attached to the PC Data Port Module, plug the appropriate end of the PCDPM-to-MDPM interface cable into the 14-pin (**J1**) MDPM connector on the PCDPM. (The other end of the cable will later be attached to the MDPM after it is installed.)

   IMPORTANT If the correct end of the cable is not connected into J1, the attached device will not function properly. Make sure that the notched side of the cable's plastic connector faces away from the PCDPM board, and the smooth side lies flat against it.
5. Carefully thread the cables connected to the PCDPM through the slots on the access panel. Re-attach the access panel to the phone.
6. Secure the access panel to the PCDPM and the phone using the mounting screw supplied in the PCDPM kit.
7. *Because a Modem Data Port Module will be attached to the PC Data Port Module,* see the instructions on **page 356**. The only purpose of installing a PCDPM in a digital telephone is to provide the copper path between the MDPM and a PKM (DSS/BLF) unit. There is no need to install a PCDPM without also installing an MDPM.

Figure 9: PCDPM Installation in an Executive or Standard Display Phone



Older PCDPMs, part number 550.3014, require AC transformers when they are connected to DSS/BLF units only. PCDPMs do not require AC transformers when they are used for off-hook voice announce only or have attached MDPMs. For a diagram of AC transformer requirements, see Figure 8 on **page 348**. Also, some early PCDPM-to-MDPM cables may have the stripe on the wrong edge. If the MDPM does not respond properly, try reversing the connector on the PCDPM.

Figure 10: PCDPM Cables on Executive or Standard Display Phones

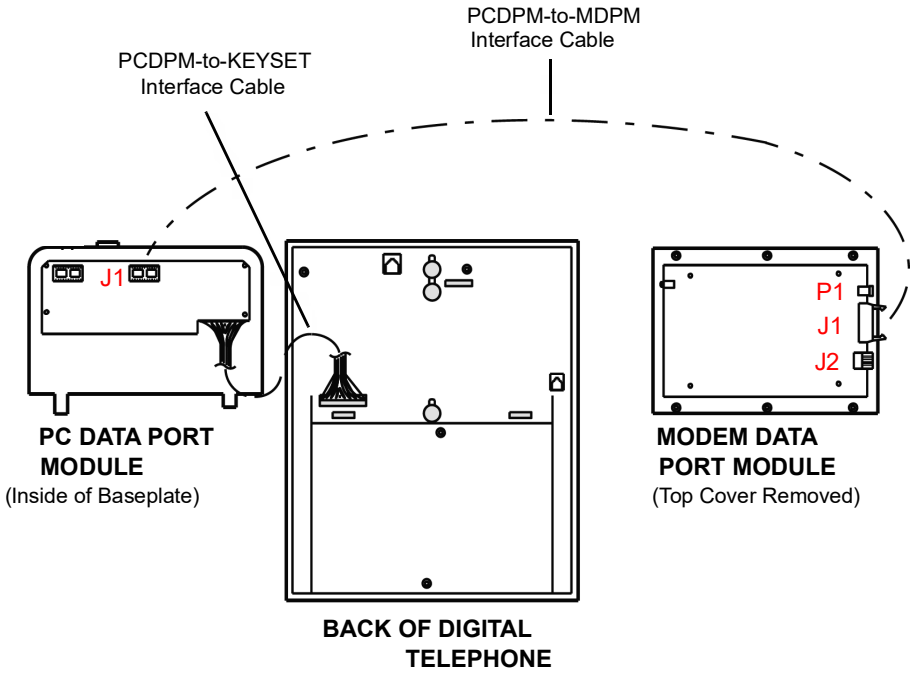


Figure 11: PCDPM Cables on Professional or Associate Display Phones

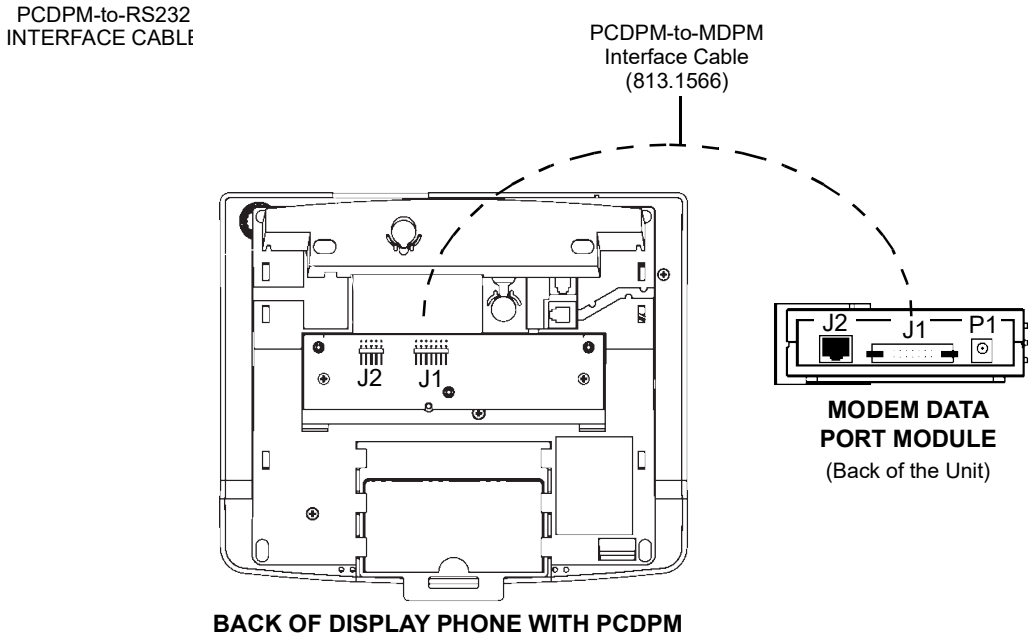


Figure 12: PCDPM Installation in a Mitel 8520 or 8560 Phone

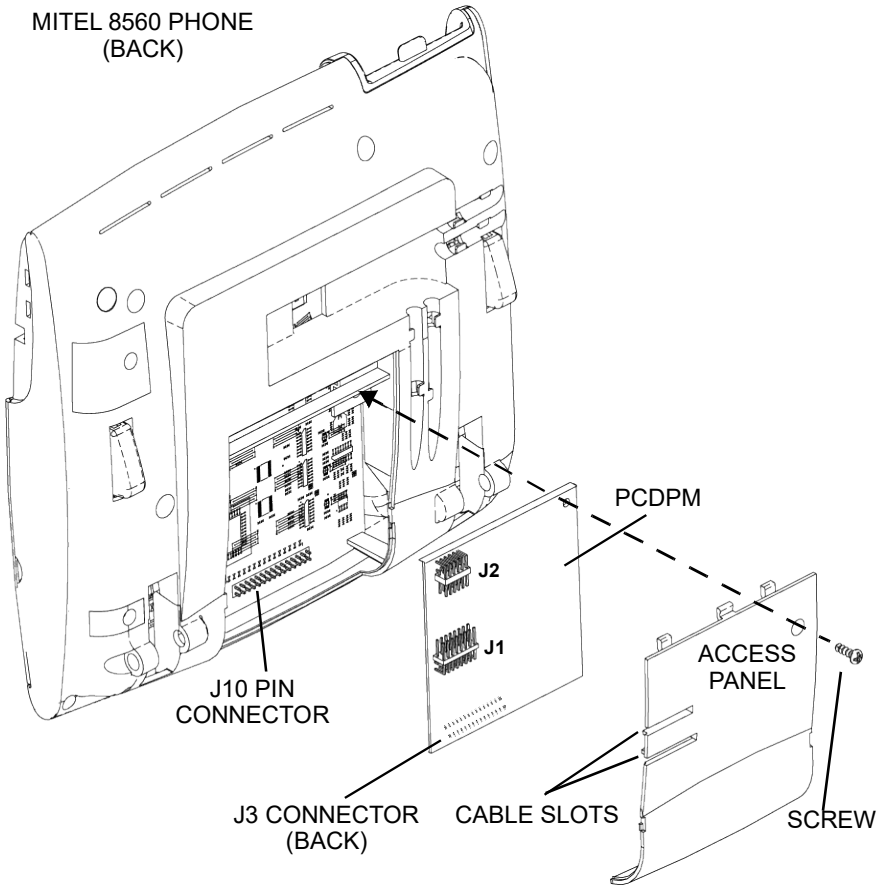
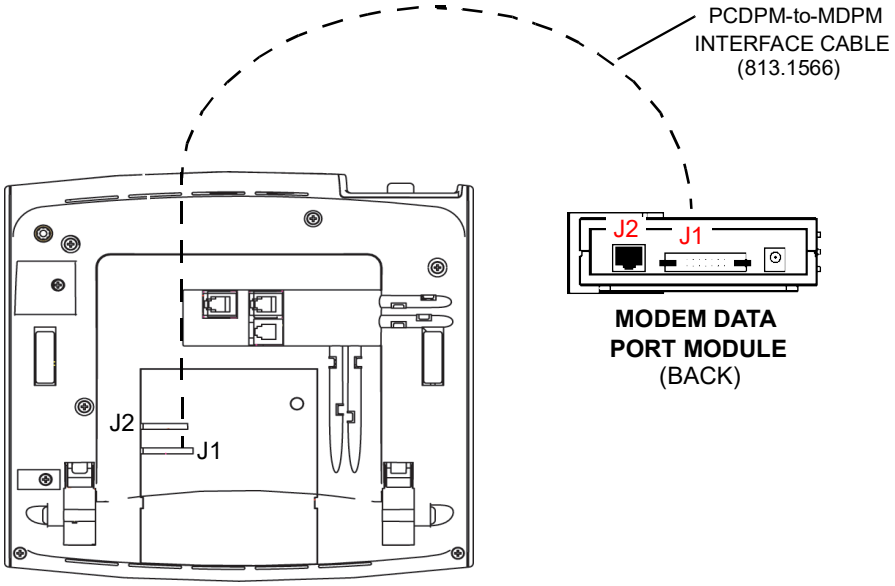


Figure 13: MDPM Cable Connection in a Mitel 8520 or 8560 Phone



Installing an MDPM

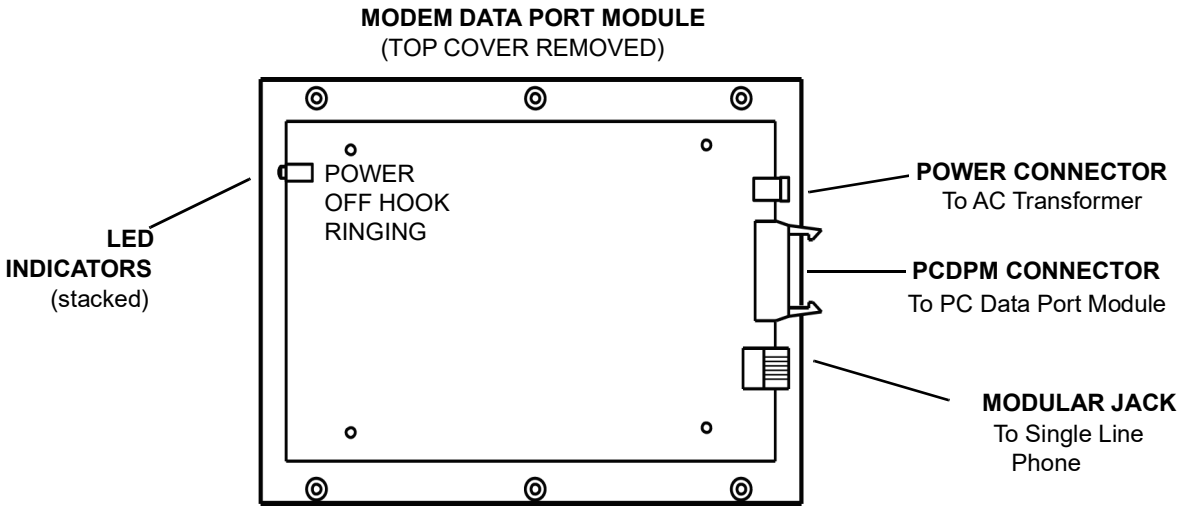
Digital telephones equipped with PC Data Port Modules may be equipped with Modem Data Port Modules, part number 550.3015 [PB2549-A in Europe]. The MDPM contains a four-conductor modular jack [line jack unit] that can be used to connect a single line phone or trunk to the digital telephone. Each “equipped” MDPM installed counts as one single line device.

For installation diagrams, see Figure 10 on **page 354** and Figure 14 on **page 357**.

To install the MDPM:

1. Ensure the phone line cord is unplugged from its modular jack on the phone.
2. Plug the free end of the PCDPM-to-MDPM interface cable into connector **J1** on the MDPM. The other end of the cable was previously attached to the PCDPM (see **page 360**).
3. Insert the phone line cord, which would normally be connected to a CO jack, into the modular jack (**J2**) on the MDPM.
4. Plug the main AC transformer unit into an AC power source. *Do not* use the same AC outlet used for the equipment chassis.
5. Reconnect the phone line cord. The MDPM can be placed flat or standing on-end.

Figure 14: Digital Telephone Modem Data Port Module (MDPM) Installation



Connecting DSS/BLF Units

Up to 20 DSS/BLF Units can be installed on the system. A single phone can be equipped with up to four units.

Before installing DSS/BLF units on digital telephones, each phone must be equipped with a PCDPM and an MDPM. For detailed installation instructions, see “Installing an MDPM” on **page 1-356** and “Installing a PCDPM” on **page 1-346**.

Connecting PCDPM Cables

For OHVA receive capability and to allow an optional DSS/BLF Unit and/or an optional MDPM to be installed, any 2-line or 6-line display phone may be equipped with a PCDPM.

- For Executive Display and Standard Display phones, use PCDPM part number 550.3018 [PB2585 in Europe]—an external AC transformer is *not* required.
  Or, use part number 550.3014 [PB2548-A]—an external AC transformer is required.
- For Professional Display and Associate Display phones, use PCDPM part number 550.3018 An external AC transformer is *not* required.
- For the Mitel 8520 and 8560 phones, use PCDPM part number 550.8400. An external AC transformer is not required.

For details, see PCDPM installation instructions included with the unit.

To Executive, Standard, Professional, or Associate Display Phones

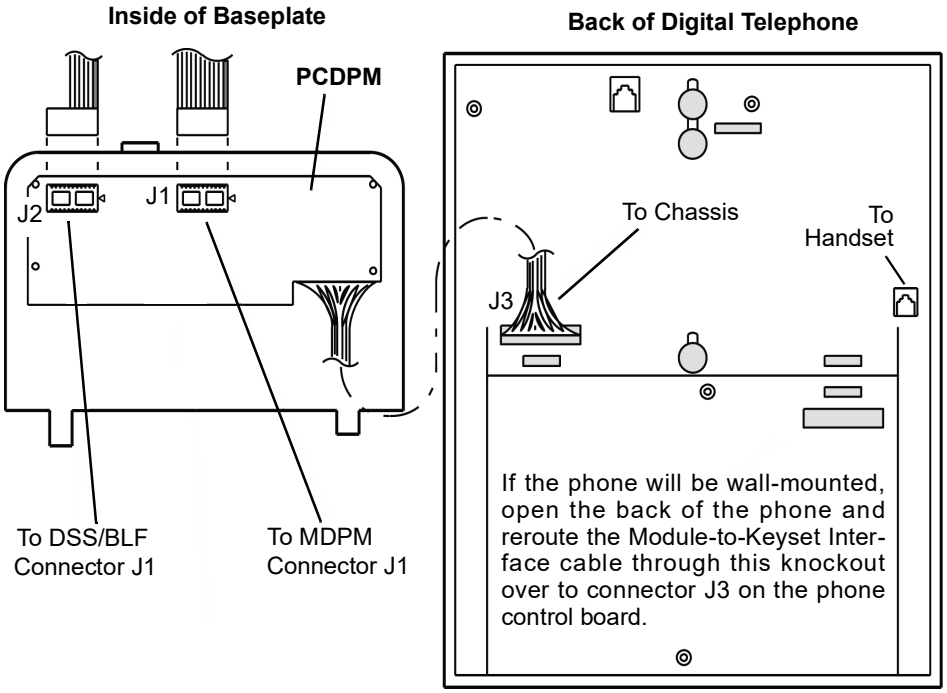
To connect the PCDPM cable:

1. Install the PCDPM for an Executive, Standard, Professional, or Associate Display phone.

   For installation diagrams, see Figure 15 on **page 358**, Figure 17 on **page 360**, and Figure 18 on **page 361**.

2. Attach one end of the DSS-to-DSS interface cable, part number 813.1519 which is included with the DSS/BLF Unit, to **J2** on the PCDPM. See Figure 15 on **page 358**. The other end of the cable will be attached to the DSS/BLF Unit after it is installed. If using an older PCDPM, part number 550.3014, plug the main AC transformer into an available AC power source. Do *not* use the outlet for the equipment chassis. Newer PCDPMs do *not* require an external AC power source.

Figure 15: PCDPM Installation on Executive or Standard Display Phones



Older PCDPMs, part number 550.3014, require AC transformers when they are connected to DSS/BLF units only or to PC/output devices only. PCDPMs do not require AC transformers when they are used for OHVA only or have attached MDPMs. For a diagram of AC transformer requirements, see the **Specifications** chapter. Also, some early PCDPM-to-MDPM cables may have the stripe on the wrong edge. If the MDPM does not respond properly, try reversing the connector on the PCDPM.

Figure 16: Back of 60-Key, 6-Row Digital DSS/BLF Unit

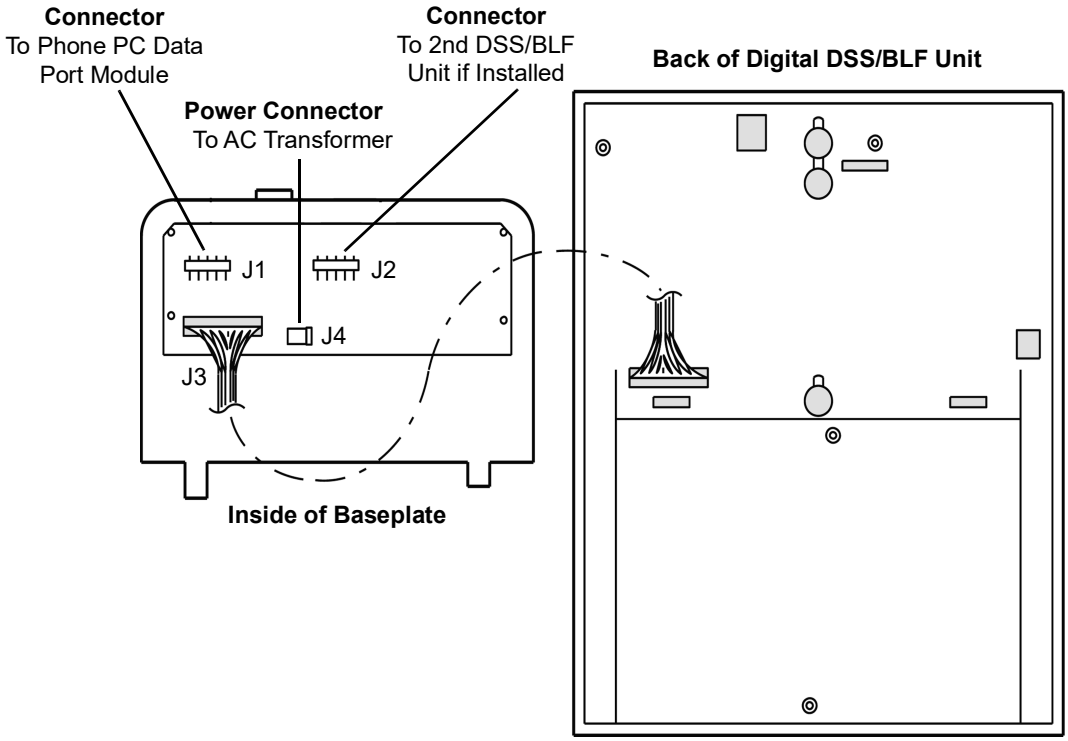


Figure 17: Executive or Standard Display Phone PCDPM Cable Connections

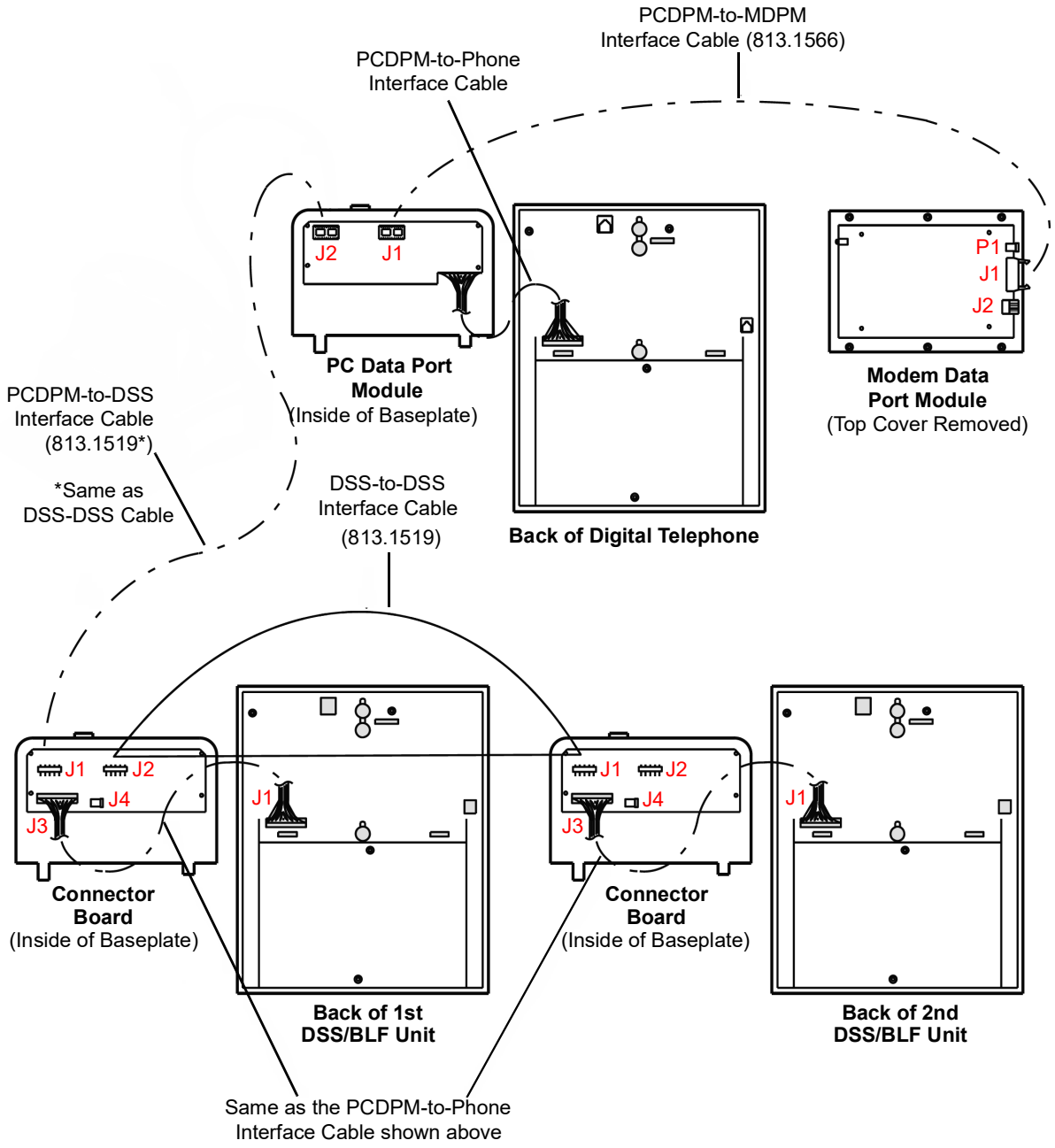


Figure 18: PCDPM Cables on Professional or Associate Display Phones

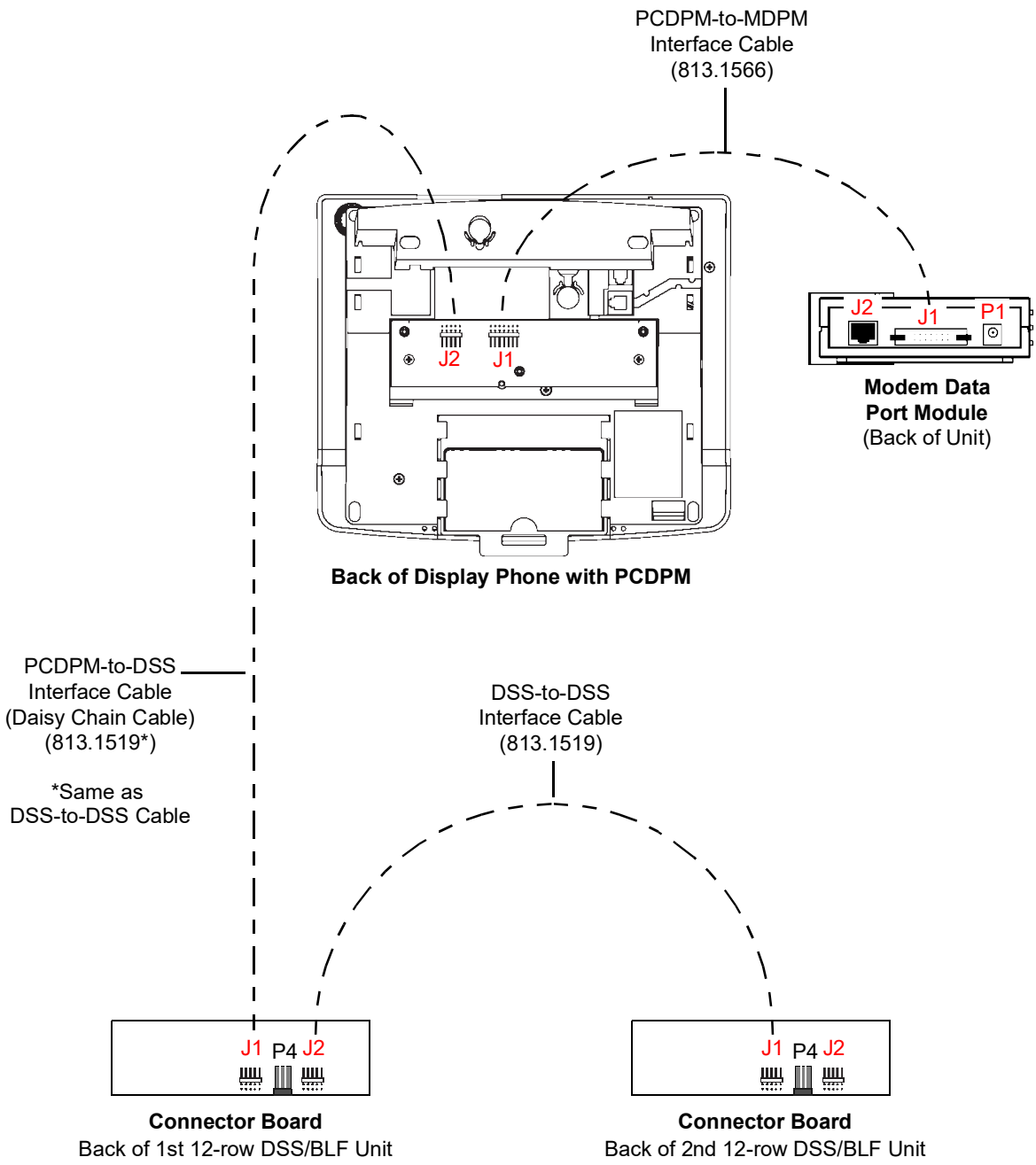


Figure 19: Mitel 8520 or 8560 Phone PCDPM Installation

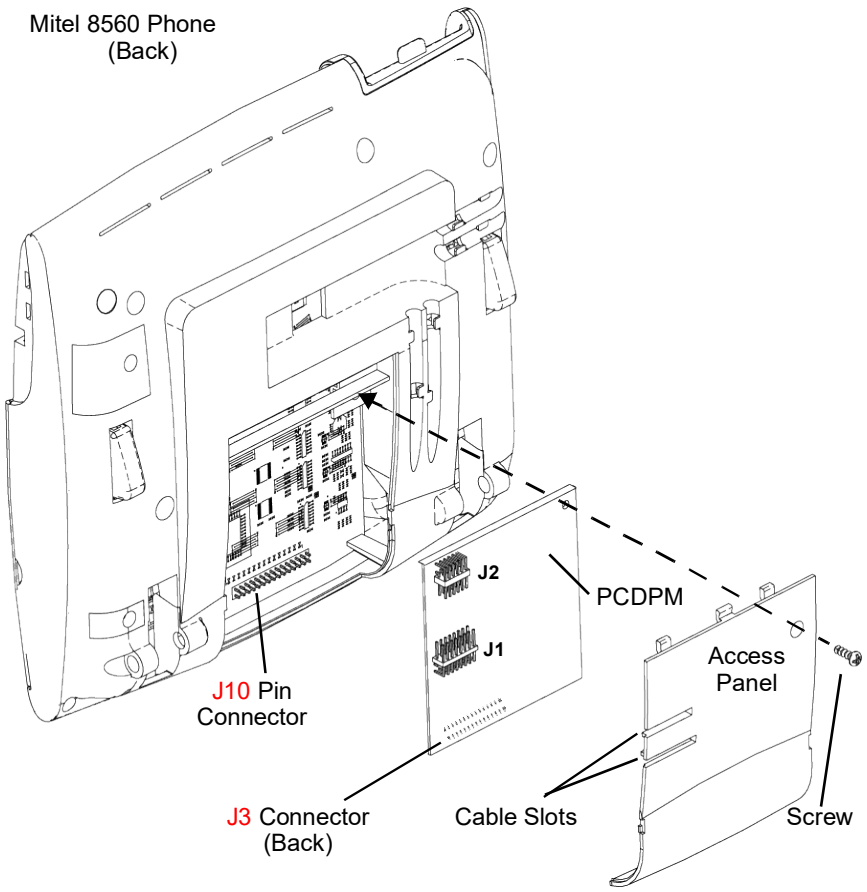
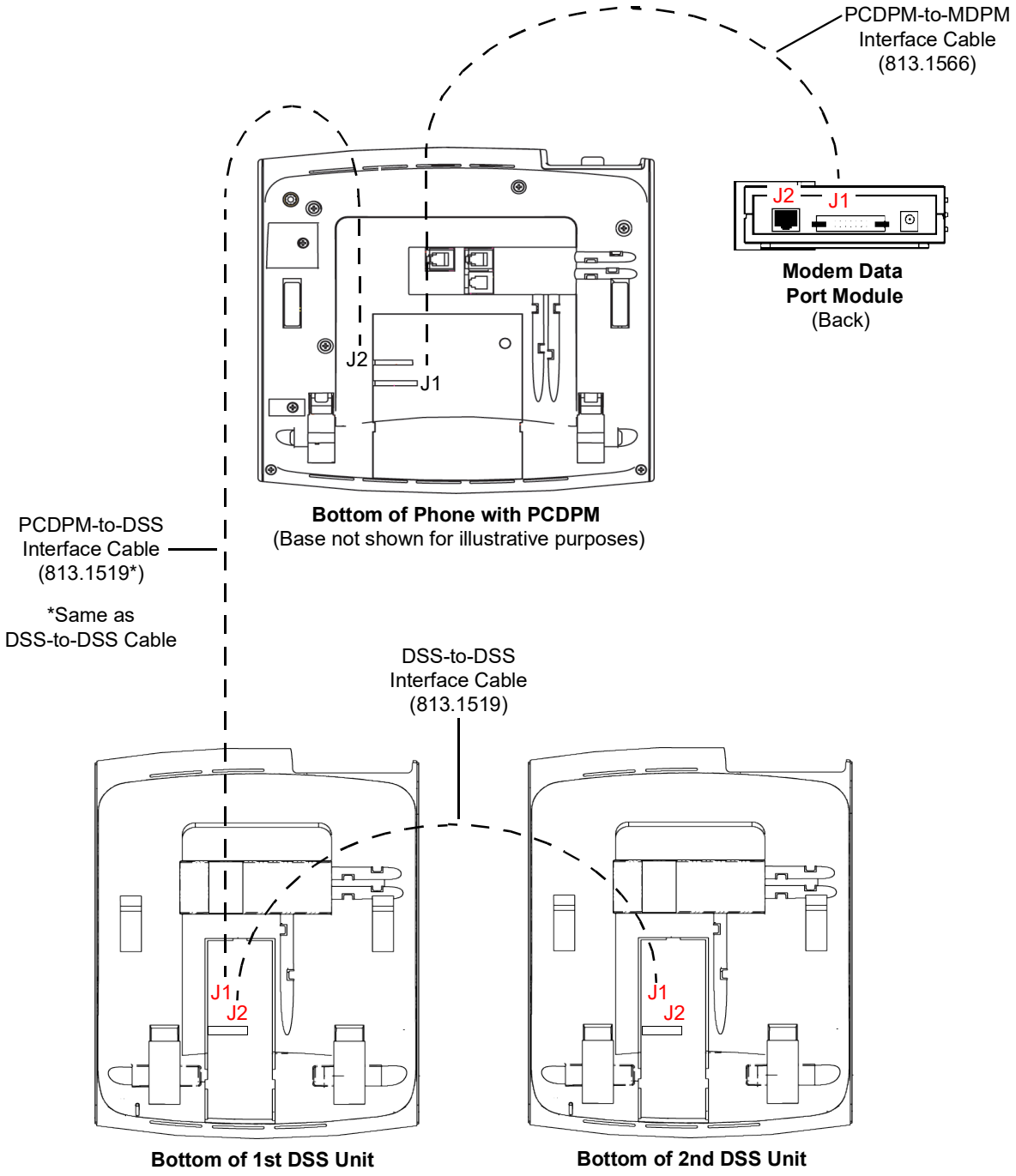


Figure 20: Mitel 8520 or 8560 Phone PCDPM Cable Connections



Connecting to Mitel 8520 or 8560 Phones

To connect PCDPM cable:

1. Install the PC Data Port Module for a Mitel 8520 or 8560 phone. For installation diagrams, see Figure 19 on **page 362** and Figure 20 on **page 363**.
2. Do either of the following:
   - *If a DSS/BLF Unit will be attached* to the PC Data Port Module, attach one end of a PCDPM-to-DSS interface cable to the 10-pin (**J2**) connector on the module, as shown in Figure 19 on **page 362**. The other end of the cable will later be attached to the DSS/BLF Unit after it is installed.

     IMPORTANT If the correct end of the cable is not connected into J2, the attached device will not function properly. Make sure that the notched side of the plastic connector on the cable is facing away from the PCDPM board and the smooth side lies flat against it.

   - *If a Modem Data Port Module will be attached* to the PC Data Port Module, plug the appropriate end of the PCDPM-to-MDPM interface cable into the 14-pin (**J1**) MDPM connector on the PCDPM. The other end of the cable will later be attached to the MDPM after it is installed.

     IMPORTANT If the correct end of the cable is not connected into J1, the attached device will not function properly. Make sure that the notched side of the plastic connector on the cable faces away from the PCDPM board and the smooth side lies flat against it.

3. Carefully thread the cables connected to the PCDPM through the slots on the access panel. Re-attach the access panel to the phone.
4. Secure the access panel to the PCDPM and the phone using the mounting screw supplied in the PCDPM kit.
5. Do either of the following:
   - *If a Modem Data Port Module will be attached to the PCDPM,* see “Installing an MDPM” on **page 1-356**.
   - *If a Modem Data Port Module will **not** be attached to the PCDPM*, reconnect the line cord.

Attaching 6-Row and 50-Key DSS Units

Install the 6-Row or Mitel 8450 50-Key (50-Button) DSS/BLF Unit as outlined in the following procedure. For diagrams of the installation, see Figure 17 on **page 360** and Figure 16 on **page 359**.

1. Remove the baseplate from the back of the DSS/BLF Unit.
2. On the connector board located in the baseplate, plug the free end of the PCDPM-to-DSS interface cable into the main connector (**J1**). The other end of the cable was previously attached to the PC Data Port Module (PCDPM) on the corresponding phone.
3. Plug the AC transformer cable into the power connector (J4) on the baseplate connector board. The transformer will later be plugged into an AC power source. DSS/BLF Units always require separate AC transformers. For a diagram of AC transformer requirements, see Figure 8 on **page 348**.

4. To attach a second DSS/BLF Unit to the first unit:
   a. Plug one end of a DSS-to-DSS interface cable into the secondary connector (**J2**) on the baseplate connector board in the first unit.
   b. Remove the baseplate from the back of the second DSS/BLF Unit.
   c. Plug the other end of the DSS-to-DSS interface cable into the main connector (**J1**) on the baseplate connector board in the second unit.
   d. Plug the AC transformer cable into the power connector (**J4**) on the baseplate connector board in the second unit. The transformer will later be plugged into an AC power source. If more than one DSS/BLF unit is installed on a phone, each unit must be equipped with an AC transformer. In addition, if using PCDPM part number 550.3014 on the corresponding phone, the PCDPM also must be equipped with an AC transformer. PCDPM part number 550.3018 does not require an AC transformer. For details, see "Connecting PCDPM Cables" on **page 1-357**.
5. To attach a third unit to the second unit, follow the same substeps outlined in step **4**.
6. Perform the following DSS/BLF Unit self-test:
   a. While holding down the first two buttons in row one (extension numbers 1000 and 1001 in the default database), plug the main AC transformer unit into an available AC power source. Do ***not*** use the outlet for the system chassis. Check that all buttons light. If an LED does not light, unplug the AC transformer and plug it in while holding down the first two buttons in row one (ext. 1000 and 1001). If the test fails again, replace the DSS/BLF Unit.
   b. Press each button individually (in any order) to remove the lit status. When finished, all buttons should not be lit. If an LED does not go out, unplug the AC transformer and repeat steps **a** and **b**. If the test fails again, replace the DSS/BLF Unit. After all of the buttons have been tested, it takes approximately 5 seconds for the DSS/BLF Unit to return to normal operation.
   c. Replace the baseplate on the back of the DSS/BLF Unit, taking care not to snag any cables between the baseplate and housing.

Attaching a 12-Row DSS Unit

Install each 12-row DSS/BLF Unit as outlined following procedure. For diagrams of the installation, see Figure 18 on **page 361**. A description of the units appears in "Direct Station Select/Busy Lamp Field Units" on **page 1-116**.

1. Attach one end of a PCDPM-to-DSS interface cable, part number 813.1519, into connector **J2** on the PCDPM.
2. Remove the baseplate on the back of the DSS/BLF console.
3. On the connector board located in the baseplate, plug the free end of the PCDPM-to-DSS interface cable, part number 813.1519, into the main connector (**J1**).
4. Plug the AC transformer cable into the power connector (**P1**) on the baseplate connector board. The transformer will later be plugged into an AC power source. DSS/BLF Units always require their own separate AC transformers. For a diagram of AC transformer requirements, see Figure 8 on **page 348**.
5. To attach a second 12-row DSS/BLF Unit to the first unit:

a. Plug one end of a DSS-to-DSS interface cable, part number 813.1519, into the secondary connector (**J2**) on the baseplate connector board in the first unit.

b. Remove the baseplate from the back of the second DSS/BLF Unit.

c. Plug the other end of the DSS-to-DSS interface cable into the main connector (**J1**) on the baseplate connector board in the second unit.

d. Plug the AC transformer cable into the power connector (**P1**) on the baseplate connector board in the second unit. The transformer will later be plugged into an AC power source. If more than one 12-row DSS/BLF Unit is installed on a phone, each unit must be equipped with an AC transformer. In addition, the PCDPM on the corresponding phone must be equipped with an AC transformer.

6. To attach a third unit to the second unit, follow the same substeps outlined in **step 5**.

7. Perform the 12-row DSS/BLF Unit self-test:

a. While holding down the first two buttons in row one (extension numbers 1000 and 1001 in the default database), plug the main AC transformer into an available AC power source. Do ***not*** use the outlet for the system chassis. Check that all of the buttons light. If an LED does not light, unplug the AC transformer and plug it in while holding down the first two buttons in row one (ext. 1000 and 1001). If the test fails again, replace the DSS/BLF Unit.

b. Press each button individually (in any order) to remove the lit status. When finished, check that all buttons are unlit. If an LED does not go out, unplug the AC transformer and repeat steps **a** and **b**. If the test fails again, replace the DSS/BLF Unit. After all buttons have been tested, it takes approximately five seconds for the DSS/BLF Unit to return to normal operation.

c. Replace the baseplate on the back of the DSS/BLF Unit, taking care not to snag any cables between the baseplate and housing.

Installing a Single Line Handset Amplifier

Single line set users may want to install a handset amplifier. Usually, the amplifier is an external device placed between a single line phone and the handset. For specifications, see **page 120**.

To install the handset amplifier:

1. Unplug the coiled handset cord from the single line set.
2. Plug the coiled handset cord into the amplifier jack labeled HANDSET.
3. Plug the amplifier line cord (coming from the jack labeled TELEPHONE) into the single line set handset jack.
4. Plug the amplifier power supply cord into an available AC power source. *Do not* plug the transformer unit into the outlet used for the Mitel 5000 chassis. Also, Mitel recommends using a UPS.
5. Increase or decrease the handset volume, using the thumbwheel located on the amplifier. Verify that the amplifier is functioning correctly by placing a call and adjusting the volume from low to high.

Chapter 6

Reference

Chapter Overview

The Reference chapter contains information such as technical support contact information and replacement parts and part numbers. Phone feature codes as well as system-wide feature codes are listed. Acronyms and glossary terms are also provided.

Technical Support Contact Information

Table 57 summarizes important contact information you may need when installing, upgrading, or servicing a MiVoice Office 250. For your reference, write in the applicable e-mail address and phone number.

Table 57: Contact Information

Department or Function	E-mail Address	Phone No.
Local Mitel provider (provide at right)		
Technical Publications	Tech_Pubs@Mitel.com	N/A
Technical Support Department	N/A	1-888-777-3279

Replacement Parts

This section provides information to help the customer order and stock replacement parts for the MiVoice Office 250.

Parts Ordering Procedure

When ordering equipment for the MiVoice Office 250, provide the following information to the order processing clerk:

- Company name
- Purchase order number
- Required date of shipment
- Part number(s) of equipment ordered
- Quantity required

Rationale for Stocking Spare Parts

Mitel is dedicated to providing and supporting the best possible service for all customers. As a way of achieving this goal, Mitel recommends that its providers establish and maintain an inventory of spare parts to minimize interruption of service to customers' business operations.

To minimize the effects of any service outage that a customer may experience, critical parts should be kept as spares in dealer and branch inventory. However, because each customer's installation may be different, coming up with a meaningful list of spare parts that would apply to all customers is not feasible. Mitel recommends that providers assess the critical needs of their installed customer base.

Providers need to identify the parts that, in the event of system failure, would result in a significant interruption of the customer's business operations. Having been identified for each customer's needs, these critical parts could be considered a meaningful list of spare parts to keep in inventory. The spare parts inventory should be adjusted in accordance with changes that occur to the customer base or to the versions of items being maintained in inventory. Customers may join or leave the installed base, or existing customers may modify their systems or purchase new equipment. As well, changes to hardware, firmware, and software may affect the currency or effectivity of the revision levels being maintained in the dealer or branch spare parts inventory.

Parts and Part Numbers

For reference purposes only, the following tables list selected MiVoice Office 250 equipment, software, licenses, and documentation. The part numbers are valid and the descriptions coincide with the names used in the manual. However, the names shown may differ from the names appearing on the current Mitel Parts List. For ordering purposes, use the following tables for reference but rely on the current Mitel Price List for details.

Version 5.0 Parts

The following table lists new parts and part numbers for v5.0.

Table 58: Version 6.2 or later Parts and Part Numbers

Part	Part Number
System Software	
MiVoice Office 250 v4.0 to v6.0 Upgrade License	54005356
MiVoice Office 250 v6.0 North America DVD	50006491
MiVoice Office 250 v6.0 United Kingdom DVD	50006492
Software Feature License	
Meet-Me Conferencing	54005359
UVM E-Mail Synchronization — Enhanced Integration	54005357
UVM E-Mail Synchronization — Blackberry Enhanced Integration	54005358
Hot Desking	54005399
Hardware	
One GB Serialized Compact Flash-Type Card (North America)	841.1152
Two GB Serialized Compact Flash-Type Card (North America)	50006500
One GB Serialized Compact Flash-Type Card (UK)	841.1154
Two GB Serialized Compact Flash-Type Card (UK)	50006501
Documentation	
MiVoice Office 250 Phone User Guide Supplement	835.3262-3
UVM E-Mail Synchronization Administrator Guide	835.3286
UVM E-Mail Synchronization User Guide	835.3287

Replacement Parts

Table 59 identifies the system equipment and parts available for use in a MiVoice Office 250.

Table 59: Replacement Parts

Description	Part Number [Europe]
HX Controller Processor Module (PM)	580.3000
Processor Expansion Card (PEC-1)	580.2020
16-Port Digital Desktop Module (DDM-16)	580.2201
DDM-16 Termination Cable	813.1910
Digital Expansion Interface (DEI) – See Table 62, “Digital Expansion Components and Licenses,” on **page 374**.	
Miscellaneous Equipment	
MiVoice Office 250 Modem Kit	550.3026
5-Button Keypad, PCBA	580.2600
Null Modem with Hardware Flow Control for Processing Server (PS-1)	804.2605
24V Power Adapter for the 8528/8568 phones (requires cord)	50005300
C7 Power Cord with North America (NA) plug type	51005172
C7 Power Cord with European plug type	51000582
	(Sheet 1 of 2)

Table 59: Replacement Parts (Continued)

Description	Part Number [Europe]
120 W External Power Supply Unit	580.9126
C13 AC Power Cord (North America)	50006271
C13 AC Power Cord (UK)	50005612
Standard Chassis and Power Supply Wall-Mount Kit	828.1771
Expanded Chassis, Power Supply, and Two DEI Wall-Mount Kit	50006504
	(Sheet 2 of 2)

Trunk Interfaces/Modules

Table 60 summarizes the modules available to connect to trunks on the public telephone network.

Table 60: Public Network Trunk Interfaces

Description	Part Number
AudioCodes MP-114 SIP Gateway Quintum AFT 400 gateway For a current list of compatible SIP gateways, refer to KB article 09-4940-00056 on the Mitel Knowledge Base Center (http://domino1.mitel.com/prodsupp/prodsupkb.nsf/WebSearchForm?OpenForm).	901.3924 590.5349
IP Single Line Adapter (IP-SLA)	770.3000
ISDN Basic Rate Module (BRM-2), Two-Port [Sold only in Europe]	580.2240
Loop Start Module, Four-Port (LSM-4)	580.2304
T1/E1 Module (T1M)	580.2700
Dual T1/E1 Module (T1M-2)	580.2702

Licenses

Table 61 summarizes the licenses available for MiVoice Office 250 systems.

Table 61: MiVoice Office 250 Licenses

Description	Part Number
IP Phones	
Category A — IP Phone	840.0226
Category B — IP Phone	840.0414
Category C — IP Phone	840.0415
Category D — IP Phone	840.0416
Category E — Digital Telephones	840.0417
Category F — SIP Phones	840.0418
Miscellaneous Licenses	
Single Node Connection License (System Manager)	840.0268
Call Processing Application Snap-In MiVoice Office 250, v1.0 License	840.0237
T1 Second Port	840.0759
IP Gateway Trunk License	840.0234

Table 61: MiVoice Office 250 Licenses(Continued)

Description	Part Number
PRI License	840.0227
License for Second Digital Expansion Interface (DEI) chassis	840.0410

Digital Expansion Equipment

Table 62 identifies the components available for equipping a MiVoice Office 250 with digital capability.

Table 62: Digital Expansion Components and Licenses

Description	Part Number
Digital Expansion Interface (DEI) chassis	580.1001
Digital Endpoint Module (DEM-16)	580.2200
Digital Endpoint Module (DEM-16) Adapter (Quad RJ-45 to Amp), 15 ft (4.5720 m)	813.1814
Extended DEI Cable (for rack-mount installations using DEIs)	50006488

Cables

Table 63 identifies cables required to install and maintain MiVoice Office 250 equipment and selected peripherals.

Table 63: MiVoice Office 250 Equipment and Peripheral Cables

Description	Part Number
Cable Assy, DB15 to 2x7 Female Twist Ribbon	813.1821
Voice Processor Expansion Cable	813.1806
Mitel 48VDC Ethernet Power Adaptor 100–240V 50–60Hz(ES)	50005301
Mitel 24VDC C7 Power Cord with U.S. Plug Type	51005172
Digital Expansion Interface (DEI) Cable, 8 in. (20.32 cm)	813.1847 or 813.1847–001
PCDPM-to-Phone Interface Cable	813.1567 [C108]
PCDPM-to-MDPM Interface Cable	813.1566 [C107]

Analog Single Line Modules

Table 64 identifies the single line modules available for equipping a MiVoice Office 250 with analog single line phones. The four-port single line module (SLM-4) resides in the MiVoice Office 250 controller. Up to four SLM-4s can be installed in the HX Controller chassis, but only one SLM-4 can be installed in the CS Controller chassis. The eight-port single line module (SLM-8) resides in a DEI bay, and a system can support up to twelve SLM-8 modules.

Table 64: Single Line Modules

Description	Part Number
Four-port Single Line Module (SLM-4)	580.2100
Eight-port Single Line Module (SLM-8)	580.2101

Batteries

Table 65 identifies the batteries that can be replaced on a MiVoice Office 250.

Table 65: Batteries

Description	Part Number
Database Back-Up Battery (3.6V, coin lithium-ion)	815.1011
Rechargeable Battery (PM-1 Only)	815.1026

Direct Station Select (DSS)/Busy Lamp Field (BLF) Equipment

Table 66 summarizes equipment, cables, and parts needed to connect the Mini-DSS Unit to IP phones and DSS/BLF units to digital telephones that operate on a MiVoice Office 250.

Table 66: DSS/BLF Units, PKMs, Data Port Modules, and Transformers

Description	Part Number [Europe]
Mitel IP PKM Kit. 12-Key (comes with one PKM IM)	50005914
Mitel IP PKM Kit. 48-Key (comes with one PKM IM)	50005915
Mitel PKM FRU. 12-Key Dark Gray	50002822
Mitel PKM FRU. 48-Key Dark Gray	50002824
Mitel PKM Interface Module IM (v2) for the 5224/5324/5330/5340/8528/8568 phones	50005662
PKM 16 (8416 16-Button Mini-DSS)	550.8416 (Discontinued)
24VDC PSU required for PKM's (Required for 8528/8568 phones)	50005300
PKM 50 (8450, 50-Button, 10-row x 5-column)	550.8450 (Discontinued)
60-Button, 12-row x 5-column	560.4000 (Discontinued)
60-Button, 6-row x 10-column	550.4200 (Discontinued)
PC Modem Data Port Module (PCDPM) Executive or Standard Display phones (does *not* require an external AC transformer)	550.3018 [PB2585] (Discontinued)
PC Modem Data Port Module (PCDPM) 8520 or 8560 phones (Does not require an external AC transformer.)	550.8400 [PB2612] (Discontinued)
Modem Data Port Module (MDPM) Executive Display, Standard Display, 8520, or 8560 phones (Includes PCDPM-to-MDPM cable and AC transformer.)	550.3015 [PB2549-A] (Discontinued)
Transformer for PCDPM and MDPM units (24 VAC,450 mA)	806.1045 (Discontinued)

Phones, Softphones, and Related Phones

Table 67 identifies supported phones, softphone applications, etc.

> **NOTE** 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

Table 67: Phones

Phone	Part Number [Europe]
IP and SIP Phones	
8600 Multi-Protocol Phone with No Display	550.8600
8602 IP Softphone Application	840.0477
8622 – 2-Line Display Multi-Protocol Phone	550.8622
8660 – 6-Line Display IP Phone	550.8660
8662 – 6-Line Display Multi-Protocol Phone	550.8662E
8664 – 2-Line Display 802.11b Wireless IP Phone	550.8664
8665 – 2-Line Display 802.11b Wireless IP Phone	550.8665
8668 – 2-Line Display 802.11b Wireless IP Phone	550.8668
8690 Multi-Protocol, Multimedia Touchscreen Phone	550.8690
Internet Protocol Single Line Adapter (IP SLA)	770.3000
5212 IP Phone	50004890
5224 IP Phone	50004894
5304 IP Phone	51011571
5312 IP Phone	50005847
5320 IP Phone	50006191
5324 IP Phone	50005664
5330 IP Phone	50005804
5340 IP Phone	50005071
5360 IP Phone	50005991
5610 Cordless Handset and IP DECT Stand (North America)	51015276
5610 Cordless Handset and IP DECT Stand (UK)	51015281
5310 IP Conference Module	50005321
5310 IP Conference Saucer (Dark Grey)	50004459
5310 IP Conference Saucer (Silver)	50004460
5310 Optional Remote Control Mouse for the 5324 IP Phone	50001543
Gigabit Ethernet Stand	51009841
Wireless LAN Stand	51009840
Line Interface Module (LIM) (Canada, Caribbean, North America, Netherlands, France, Australia, Luxembourg, Finland, Denmark, Austria)	50004197
Line Interface Module (LIM) (UK, Ireland, Italy, Belgium, Germany, New Zeland, Portugal, Spain, Norway, Sweden)	50004198
Cordless Handset and Module Bundle (NA DECT)	50005711
	(Sheet 1 of 2)

Table 67: Phones (Continued)

Phone	Part Number [Europe]
Cordless Headset and Module Bundle (NA DECT)	50005712
Cordless Accessories Module (NA DECT)	50005521
Cordless Handset with Charging Plate (NA DECT)	50005405
Cordless Headset with Charging Cradle (NA DECT)	50005522
Cordless Headset and Module Bundle (EURO DECT)	50005713
Cordless Headset and Module Bundle (EURO DECT)	50005714
Cordless Accessories Module (EURO DECT)	50005402
Cordless Handset with Charging Plate (EURO DECT)	50005403
Cordless Headset with Charging Cradle (EURO DECT)	50005404
Mitel Standard IP phone power supply	50005301
Digital Telephones	
Associate Display	560.4200
Executive Display – with red and green LEDs	550.4500 [KT536]
8500 – with red and green LEDs	550.8500 [KT540]
8520 – with red and green LEDs	550.8520 [KT539]
8560 – with red and green LEDs	550.8560 [KT538]
8528 (U.S.)	50006122
8528 (U.K.)	50006321
8568 (U.S.)	50006123
8568 (U.K.)	50006322
Professional Display	560.4300
Standard Display – with red and green LEDs	550.4400 [KT535]
Single-Line Adapter	
Single-Line Adapter	550.2500
Transformer for SLA (24 VAC, 1.25 A)	806.1046 [PS2731]
	(Sheet 2 of 2)

Documentation

Table 68 identifies the existing documentation that supports the MiVoice Office 250. For a list of new documentation created for the version 5.0 release, see Table 58 on **page 372**.

Table 68: System Documentation

Document Title	Part Number
MiVoice Office 250 Installation Manual	580.8000
MiVoice Office 250 Features and Programming Guide	580.8006
MiVoice Office 250 Installation Quick Start Guide	835.2879
Mitel System Administration & Diagnostics Guide	550.8125
MiVoice Office 250 Phone Administrator Guide	580.8001
MiVoice Office 250 UVM and EM Administrator Guide	580.8009

Table 68: System Documentation (Continued)

Document Title	Part Number
Mitel 5610 IP DECT Stand Configuration & Administration Guide	57011426
Mitel 5212/5224 Installation Guide	57007156
Mitel 5304 IP Phone Installation Guide	57009901
Mitel 5312/5324 Installation Guide	57009711
MItel 5320 IP Phone Installation Guide	57010765
Mitel 5330/5340 Installation Guide	57007717
Mitel 5360 IP Phone Installation Guide	57011154
Mitel 8528/8568 Telephone Installation Guide	835.3259
Mitel PKM 12 Installation Guide	56004531
Mitel PKM 48 Installation Guide	56004530
Mitel Wireless LAN Stand Installation Guide	000-002-765
Mitel 5310 Conference Unit Installation Guide	56006636
Mitel Line Interface Module Installation Guide	56006636
Mitel 5610 IP DECT Stand Installation Guide	57011091
Mitel 5610 RTX4002 DECT Repeater User Manual	95100805
Mitel 5610 Cordless Handset Quick Reference Guide	57011425

Table 69 identifies the existing phone user guides that are supported by the MiVoice Office 250. For a list of new documentation created for the version 5.0 release, see Table 58 on **page 372**.

Table 69: Phone User Guides

Document Title	Part Number [Europe]
IP Phone User Guides	
5212/5224 IP Phone User Guide	550.8122
5304 IP Phone User Guide	550.8034
5312/5324 IP Phone User Guide	550.8035
5320/5330/5340 IP Phone User Guide	550.8123
Mitel 5360 IP Phone User Guide	550.8124
8600 Phone User Guide	550.8115
8620/8622 Phone User Guide	550.8119
8560/8660 Phone User Guide	550.8112
8662 Phone User Guide	550.8117
8690 Phone User Guide	550.8116
Wireless Phone and Softphone User Guides	
8602 Softphone User Guide	835.3019
8664/8665/8668 Phone User Guide	935.4522
8524/8525/8526 Phone User Guide	935.4519
Digital Telephone User Guides	
8528 Telephone User Guide	550.8032
	(Sheet 1 of 2)

Table 69: Phone User Guides (Continued)

Document Title	Part Number [Europe]
8568 Telephone User Guide	550.8033
8500 User Guide	550.8114
8520 User Guide	550.8112
8560 and 8660 User Guide	550.8113
Analog Phone User Guide	
Single Line Phone User Guide	550.8107
Misc. User Guides	
Phone User Guide Supplement	835.3262
Voice Mail User Guides	
UVM and EM Voice Mail User Guide	835.3205
NuPoint UM Messaging User Guide	n/a
(Sheet 2 of 2)	

Abbreviations and Acronyms

In addition to industry-wide and/or internationally recognized units of measure, such as Hz, V, AC, DC, A, ms, etc., **Table 70** lists the abbreviations and acronyms that appear in this guide.

Table 70: Abbreviations and Acronyms

Abbreviation	Spelled-Out Meaning
ACD	Automatic Call Distribution
ADD	Automatic Diagnostics Delivery
AIS	Alarm Indication Signal
AMC	Applications Management Center
AMIS	Audio Messaging Interchange Specification
ANI	Automatic Number Identification
ANSI	American National Standards Institute
ARM	Asynchronous Response Mode
ARS	Automatic Route Selection
ASAI	Adjunct Switch Application Interface
AVDAP	Advanced Voice Data Applications Processor
AWC	Audio and Web Conferencing
AWG	American Wire Gauge
AWP	Administrative Web Portal
BLF	Busy Lamp Field
BR	Burst Rate
BRI	Basic Rate Interface
BRM	Basic Rate Module
CAT	Category
CD	Compact Disk
CF	Compact Flash (Please read the Note that follows this table.)
CID	Caller Identification
CIR	Committed Information Rate
CLID	Calling Line IDentification (British)
CLIP	Calling Line Identification Presentation (Caller ID/CLID for wireless)
CO	Central Office
CP	Call Processing
CPE	Customer Provided Equipment
CPH	Call Processing History
CRC	Cyclic Redundancy Check
CS	Communication Server
CT	Computer Telephony
DB	Database
DDI	Direct Dialing Inward (DID in U.S.)

(Sheet 1 of 5)

Table 70: Abbreviations and Acronyms (Continued)

Abbreviation	Spelled-Out Meaning
DE	Discard Eligible
DEE	Dynamic Extension Express
DEI	Digital Expansion Interface
DEM	Digital Endpoint Module
DHCP	Dynamic Host Configuration Protocol
DIN	Deutsche Institute fur Normung (German Institute for Standardization)
DKTS	A legacy reference to entities that affect digital keysets only
DLCB	Data Link Control Block
DID	Direct Inward Dialing (DDI in Britain)
DISA	Direct Inward System Access
DMI	Digital Multiplexed Interface
DMU	Diagnostics Management Utility
DND	Do-Not-Disturb
DNIS	Dialed Number Identification Service
DNS	Domain Name Server
DSL	Digital Subscriber Loop
DSP	Digital Signal Processor
DSS	Direct Station Selection
DTMF	Dual-Tone Multi-Frequency
E&M	Ear and Mouth
EMI	Electromagnetic Interference
EPROM	Erasable-Programmable Read Only Memory
ESD	Electrostatic Discharge
FCC	Federal Communications Commission
FEBE	Far End Block Error
FMR	Framer
FoIP	Fax over Internet Protocol
FPGA	Field Programmable Gate Array
FTP	File Transfer Protocol
GW	Gateway
HDB3	High Density Bipolar 3 coding
HDLC	High-level Data Link Control
HTML	Hypertext Markup Language
IC	Intercom
IE	Information element
IP	Internet Protocol
IPR	IP Resource
IPRA	IPR Application
IPRC	IP Resource Card

(Sheet 2 of 5)

Table 70: Abbreviations and Acronyms (Continued)

Abbreviation	Spelled-Out Meaning
ISDN	Integrated Services Digital Network
ITP	Inter-Tel Protocol
LAN	Local Area Network
LCD	Liquid Crystal Display
LED	Light Emitting Diode
LMC	License Management Center
LSA	Loop Start Adapter (port)
LSM	Loop Start Module
MAC	Medium Access Control (Address)
MCF	MId Call Features
MDF	Main Distribution Frame
MDPM	Modem Data Port Module
MF	Multiple Frame
MGCP	Media Gateway Control Protocol
MIME	Multipurpose Internet Mail Extensions
MIS	Management Information Systems
MOH	Music On Hold
MRA	Material Return Authorization
N/A	Not Available or Not Applicable, as appropriate
NAT	Network Address Translation
NLP	Non-Linear Process
NT	Network Terminator
NTP	Network Time Protocol
OAI	Open Architecture Interface
OLM	OnLine Monitor
OPX	Off Premises Extension
OSA	Operator System Access
OHVA	Off -Hook Voice Announce
P2P	Peer-to-Peer
PBX	Private Branch eXchange
PC	Personal Computer
PCDPM	Personal Computer Data Port Module
PCM	Pulse Code Modulation
PCR	Personal Call Routing (now called Dynamic Extension Express)
PDA	Personal Digital Assistant
PDF	Portable Document Format
PEC	Processor Expansion Card
PKM	Programmable Key Module
PM-1	Processor Module

(Sheet 3 of 5)

Table 70: Abbreviations and Acronyms (Continued)

Abbreviation	Spelled-Out Meaning
POP	Post Office Protocol
PPP	Point-to-Point Protocol
PRI	Primary Rate Interface
PS-1	Processing Server
PSTN	Public Switched Telephone Network
REN	Ringer Equivalence Number
QoS	Quality of Service
RFI	Radio Frequency Interference
RJ	Registered Jack
RMS	Root Mean Square
RTC	Real Time Clock
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transport Protocol
Si1, Si2	Names of specific bits in timeslot 0 of an E1 frame
SSH	Secure Shell
SIP	Session Initiation Protocol
SLA	Single-Line Adapter
SLM	Single Line Module
SMDR	Station Message Detail Recording
SMTP	Simple Mail Transfer Protocol
SRA	Shared Resource Allocation
SSL	Software Site License
T1M	Single-port T1/E1/PRI module
T1M-2	Dual-port T1/E1/PRI module
TCP	Transmission Control Protocol
TCR	Transceiver
TDM	Time Division Multiplex
TE	Terminal Equipment
TFTP	Trivial File Transfer Protocol
TTS	Text-to-speech
TX	Transmit
UC	Unified Communicator
UCD	Uniform Call Distribution
UDP	User Datagram Protocol
UK	United Kingdom
UM	Unified Messaging
UPS	Uninterruptible Power Supply
USA	United States of America
USB	Universal Serial Bus

(Sheet 4 of 5)

Table 70: Abbreviations and Acronyms (Continued)

Abbreviation	Spelled-Out Meaning
USOC	Uniform Service Order Code
UVM	Unified Voice Messaging
VAD	Voice Activity Detection
VoIP	Voice over Internet Protocol
VPU	Voice Processing Unit
WAN	Wide Area Network
	(Sheet 5 of 5)

Glossary

Table 71 defines terms used in this manual to describe networking and related Inter-Tel applications.

Table 71: Networking Terminology

Term	Definition
2B+D	A circuit divided into two 64 kbps B-channels that can carry voice, video, or data. It also has a 16 kbps D-channel for low-speed data and signaling. There are two types of 2B+D interfaces that use different wiring: the “U” interface uses a single twisted pair and the “S/T” interface uses two twisted pairs. This circuit is also known as Basic Rate Interface (BRI).
23B+D	A circuit divided into 23 B-channels (64 kbps each) that can carry voice, data, and video simultaneously. It also has a 64 kbps D-channel for out-of-band signaling, call control, and packet data. This interface connects with two twisted-pairs of wire. This circuit is also known as Primary Rate Interface (PRI). The Mitel system does *not* support use of the D-channel for packet data.
30B+D	The standard European ISDN interface. A 30B+D circuit is divided into 30 B-channels (64 kbps) that can carry voice, data, and video simultaneously. It also has a 64 kbps D-channel for out-of-band signaling and call control, and one framing channel. This interface uses two twisted-pairs of wire.
Automatic Call Number Identification (ANI)	Identifies the caller’s telephone number. The system receives a specified number of digits. *ANI* is another type of ANI that does not require a specified number of digits. The system receives a star (*) before the ANI digits to signal the beginning of the caller’s telephone number. Then there is another star after the digits to signal the end of the ANI information.
B-Channel	The B-channel, or “bearer” channel, is the basic component of ISDN interfaces. A B-channel can transmit or receive voice or data at up to 64,000 bits per second (64 kbps).
B-Channel Negotiation (Currently ***not*** supported on the MiVoice Office 250.)	This allows the system to request the channel that will be used for each call. When the system requests a specific B-channel for an outgoing call, the network decides which channel will be used. However, if the channel selected by the network is not the requested-channel, the system will release that channel (if seized) and attempt to re-initiate the call using the channel specified by the network. (For an incoming call, the system does need to request a channel.)
Bandwidth	The bandwidth of a device determines the frequency range it can handle. Different types of communications use different bandwidths. For example, telephone communication requires a relatively narrow bandwidth, while video requires a wide bandwidth.
Basic Rate Interface (BRI)	Also called 2B+D, BRI provides two 64 kbps B-channels that can carry voice or data and a 16 kbps D-channel for low-speed data and signaling. This can be used for homes and Centrex sites that do not require PRI.
Calling Party Number Service	Provides the calling party’s line number (not billing number) to the called party.
D-Channel	The data channel, or D-channel, carries the signaling information at 64 kbps for the B-channels in a Primary Rate Interface or at 16 kbps in a Basic Rate Interface.
	(Sheet 1 of 3)

Table 71: Networking Terminology (Continued)

Term	Definition
D-Channel Backup (Currently ***not*** supported on the MiVoice Office 250.)	Because the D-channel supports the other channels in the ISDN network, loss of the D-channel would cause loss of access to the ISDN circuits. D-channels can be backed up to provide protection in the event of D-channel loss. D-channel backup is especially important in Non-Facility Associated Signaling (NFAS), because the D-channel supports all of the B-channels in the ***entire*** network.
Dedicated Service	Dedicated service reserves a group of B-channels for a specific function, such as outgoing calls. The system uses trunk group programming to determine how the channels are used. Therefore, the system would require a separate trunk group for each function. The installer can create “PBX-provided” dedicated service on call-by-call circuits through DB Programming (such as incoming only or accessible only to a group of users) or the service provider can determine the dedicated service.
Dialed Number Identification Service (DNIS)	DNIS identifies the number that the caller dialed to reach your location. The system receives a base number and a specified number of digits that identify the dialed number. *DNIS* is a form of DNIS that does not have a specified number of digits. The system receives a star (*) before the DNIS digits to signal the beginning of the dialed number. Then there is another star after the digits to signal the end of the DNIS information.
Flexible Billing (Currently ***not*** supported on the MiVoice Office 250.)	Allows a 900 subscriber to modify a call’s billing rate while speaking with the caller.
H0 and H11 (Currently ***not*** supported on the MiVoice Office 250.)	Switched digital service types that combine contiguous B-channels into a single bit stream. H0 combines six channels into a 384 kbps bit stream. H11 combines 24 channels into a 1536 kbps bit stream. Applications for these services include video conferencing and LAN connections.
Integrated Services Digital Network (ISDN)	ISDN is a network of channels that can provide simultaneous voice, data, and video communication.
Multi-Rate ISDN (Currently ***not*** supported on the MiVoice Office 250.)	A type of switched digital service that allows the user to dial up, on demand, any number of combined B-channels.
N X 64 kbps (Currently ***not*** supported on the MiVoice Office 250.)	A switched digital service type that combines the B-channels into a single bit stream. Applications for this service include video conferencing and LAN connections.
Network Ring Again (Currently ***not*** supported on the MiVoice Office 250.)	Similar to the system queue feature that allows a caller to queue onto a busy phone and then be called when the phone is available. However, Network Ring Again allows queuing between two PBXs that are on the same ISDN network.
Non-Facility Associated Signaling (NFAS) (Currently ***not*** supported on the MiVoice Office 250.)	When a site uses several PRI circuits, the D-channel on one circuit can support the B-channels on other circuits, providing one additional B-channel per circuit (that is, 24 B-channels instead of 23B+D). This arrangement requires D-channel backup.
On-Demand B-Channel Selection	This feature allows on-demand allocation of B-channels. With on-demand B-channel selection, all B-channel trunks can be assigned to one trunk group and then programmed to be used as needed (ARS, outgoing access, etc.). This feature takes full advantage of the Mitel system features (trunk groups, ARS, and call routing) and allows maximum on-demand use of the bandwidth. This can be combined with Dedicated Service.
	(Sheet 2 of 3)

Table 71: Networking Terminology (Continued)

Term	Definition
Primary Rate Interface (PRI)	The ISDN circuit that provides B-channels that can carry voice, data, and video simultaneously. It also has one D-channel for out-of-band signaling and packet data. See also 23B+D and 30B+D. The MiVoice Office 250 does *not* support the use of the D-channel for packet data.
Primary Rate Interface Nodal/Centrex Express (Currently ***not*** supported on the MiVoice Office 250.)	Allows communication between a private branch exchange (PBX) system and a Centrex system.
Private Line Integration (Currently ***not*** supported on the MiVoice Office 250.)	Allows the system to have private lines using individual channels. The D-channel is not used on the private lines.
Span-Side Echo Canceller	The Span-Side Echo Canceller cancels out echo originating from the trunk, such as echo from the telco.
Switched Digital Services (Currently ***not*** supported on the MiVoice Office 250.)	Provide ISDN over a single telephone line by combining several B-channels into a single bit stream. Switched digital service types include H0, H11, Multi-Rate, and NX 64 kbps.
System-Side Echo Canceller	The System-Side Echo Canceller cancels out echo originating from the system, such as from a single line phone or the handset of a digital telephone.
Two-Stage Caller Identification (*ANI*DNIS*)	A two-stage address service that provides both the caller's telephone number and the dialed number. It combines the ANI and the DNIS features previously described.
User-to-User Information (UUI) (Currently ***not*** supported on the MiVoice Office 250.)	An ISDN service that allows two users to share and manipulate data information over the D-channel while maintaining a voice connection on a B-channel. For example, both parties can view the same document, discuss it, edit it and see the changes as they are made.
Wide Area Automatic Call Distribution (ACD) (Currently ***not*** supported on the MiVoice Office 250.)	Similar to the system ACD feature, this service can distribute calls to provide faster service. However, Wide Area ACD allows the calls to be sent to other PBXs in the network.
	(Sheet 3 of 3)

Appendix A

Private Networking

Introduction

The MiVoice Office 250 is capable of connecting with other Mitel platforms to form a network that supports many call processing features. This networking capability can be implemented using point-to-point T1/E1 PRI circuits, or it can be implemented using IP networking.

To the user, the network appears as though it is one integrated system. With few exceptions, users can perform all of the functions across the network that they can perform within a stand-alone telephone system.

In this appendix, all references to a “network” indicate two or more MiVoice Office 250 systems connected by a network span. Each system in a network is called a “node.” There can be up to 99 nodes per network with a combined total of up to 40,000 devices on the nodes. See the **Specifications** and **Product Description** chapters for additional information on capacities. See **Network IP Topology** for information on network topology, including firewalls and NAT.

The system supports several private networking configurations. Between any two networked systems, there can be any combination of T1/PRI or E1/PRI modules.

The Networking Wizard allows you to quickly configure the system for IP Networking or T1/E1 PRI Networking. For additional information, refer to the *MiVoice Office 250 Features and Programming Guide*.

- **T1/PRI or E1/PRI Connections**: T1M or T1M-2 modules configured this way are used for connecting system nodes in a network. Each network span requires two T1M or T1M-2 modules, one for each end of the span. See **page 395** for more details.
- **IP Networking**: DB Programming provides private networking using an IP network. See page 397 for more details. See the **Specifications** chapter for information on IP resource capacities.

Figure 4 illustrates an example of private networking using T1M or T1M-2 modules and IP resources in the MiVoice Office 250.

Figure 4: A Private Network Combining IP Resources and T1M or T1M-2 Modules

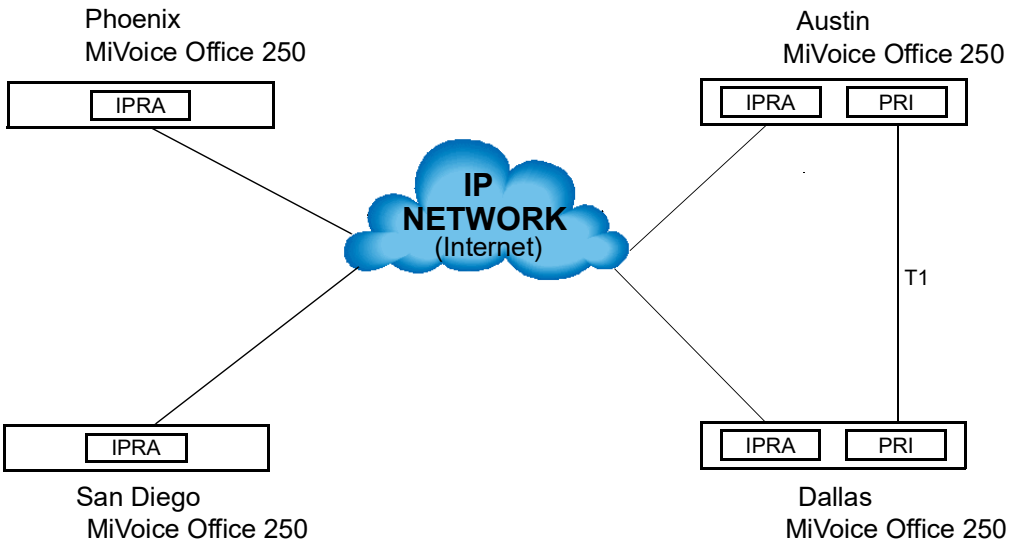


Figure 4 on **page 391** shows the networking types: IP and T1/PRI connections. Each type can communicate directly with another node that uses the same connection type. For example:

- Phoenix and Austin can speak directly to each other through the IP Resource Application and the IP network.
- Austin and Dallas can speak directly to each other through the T1 connection (PRI).
- San Diego and Dallas can speak directly to each other through the IP Resource application and the IP Network.

Nodes with different networking types can also communicate with each other, but not directly. To communicate with a dissimilar node, the call must be routed through a node with the same networking type. For example, if Phoenix needs to call Dallas, the call must first go through Austin. For Austin to speak with San Diego, the call must first go through Dallas. With proper planning and design of call routing patterns, an IP private network can be optimized such that no more than one hop is required between dissimilar nodes.

IMPORTANT	You must have sufficient PRI licenses, part number 840.0227, to configure the necessary ports on the T1M or T1M-2 module for PRI use.

Planning the Network

The most important step in setting up a network is planning. Without careful planning, the network will be difficult to program and maintain. If necessary, see **Network IP Topology** for information on network topology, including firewalls and NAT.

Gather Information

As you read through this section, fill out the planning sheet on **page 413** as you answer these questions:

- How many systems (nodes) are in the network? How will they be numbered?
- Where are they located?
- What additional hardware will you need?
- *(For PRI connections)* What additional services need to be ordered? Does the local telephone company offer point-to-point clear channel connections to the areas for the desired nodes? Does the customer have existing cable/fiber/microwave, etc., connections that can be used? In many cases, the network connection can simply be a customer-provided cable. The maximum length of the customer-provided network cable should not exceed 6000 ft (1800 m). If more than 6000 ft (1800 m) of cable is required, some type of amplification, such as a repeater, is required.
- Does the customer already have a data network? It is very possible the customer already has an existing data network. Is it possible to use bandwidth from the existing data network for voice? Does the customer want to combine voice and data on the network? What type of drop and insert devices will be included in the network plan?

NOTE For PRI connections, the System requires T1 channel 24 in any network span.

- How will the network be configured? How will the trunks connect to the network? Will all the trunks connect to one node or will the trunks be distributed?
- Call patterns for each node should be determined to configure the appropriate amount of channels necessary for the network connections. Remember to include incoming/outgoing intercom, incoming/outgoing central office, automated attendant, operator, voice mail, hunt group announcement and overflow messages, and Record-A-Call traffic in your calculations.
- What is the numbering plan for the extensions, Voice Processing applications, feature codes, hunt groups, node trunk groups, etc.? Remember to leave room for expansion in your numbering plan.
- Does each node have a local trunk group for emergency calls?
- Which system in the network will have the external voice processing system?
- How will calls ring in to the external voice processing system? EM or NuPoint UM? Will they be routed to other nodes?
- Will call routing tables and/or ARS route calls to other nodes?
- Which phone(s) will be the network-wide primary attendant(s)? Which phones are the node attendants?

- Will SMDR include all network calls? Where will the output port(s) be located? SMDR must be buffered using an external storage device.
- Where will the message print be located?
- Where will fax machines, modems, and external paging be located?
- Will there be T.38 IP faxing between MiVoice Office 250 systems. IP faxing consumes two IP resources per fax.

Design the Network

For optimum network operation, follow these guidelines:

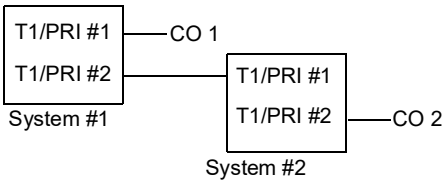
- **Carefully manage the traffic on a central node**: The central node is susceptible to unintentional bottlenecks. Tips for preventing bottlenecks on the central node are:
  - If the node contains an EM voice mail system with more than 16 ports and more than 128 Executive Display, Professional Display, or 8560 phones, redistribute the load by moving some of the Executive Display, Professional Display, or 8560 phones to another node, creating a new node, if necessary.
  - If a networked node with an EM with Automated Attendant and/or Call Routing Applications is expected to handle more than 1000 calls in an hour, use a separate EM to provide the announcements.
  - If a node with a centralized voice mail supports more than four nodes or a total of 1500 off-node devices, redistribute the load by reducing the number of phones on the central node to 200 (adding another node, if necessary).
  - A node used as an outgoing call center should not have more than 100 agents. Redistribute the load among several nodes or place the call center on its own node.
  - A node should not have more than 3 ISDN Primary Rate trunks for private networking or to the central office. Redistribute the load by moving ISDN trunks to another node, adding a node, if necessary.
  - For very busy systems or large networks, equip the satellite nodes with their own local and long distance trunks where possible.
  - If the central node is used for all long distance calls, make sure it is equipped with enough trunks to handle the peak traffic.
- **Limit the maximum number of display phones to 206 per node**: Redistribute the load by adding another node if necessary.
- **Eliminate data errors whenever possible**: Currently, the system can experience slowdowns during error dumps. The most common preventable errors are:
  - ***T1 errors***
  - ***Fiber errors***
  - Glare: Ensure the Mitel system and central office are programmed to minimize glare by initiating outgoing calls on the opposite end of a trunk group from the starting point for incoming calls. Even when configuring in this manner, glare is possible when the system gets very busy, but the likelihood of a glare condition is greatly reduced.

- **Turn off diagnostics** except when working with Engineering or Technical Support to trouble-shoot a problem. Diagnostics require processor resources and they can adversely affect the speed of heavily loaded nodes.
- **Ensure that enough T1 spans are installed between nodes to handle the traffic**: In most cases, a single T1 span between nodes will suffice. However, if increased call capacity anticipated between two nodes, add another T1 span to handle the traffic.
- **Avoid doing DB Programming during peak business hours.**
- **Avoid using System OAI on a heavily loaded node**: Redistribute the load by moving traffic not associated with the System OAI application to another node (adding a node, if necessary).
- **When using Paging**: Avoid using a lighted PKM or DSS/BLF button for the main page zone—for example, 9600 or 9601.
- Limit page zones to 60 or fewer phones for customers who need to make more than 30 pages per hour. Use external paging whenever possible.
- **Avoid overloading all-ring hunt groups**: Avoid all-ring hunt groups with 10 or more phones that are expected to handle more than 500 calls in an hour. Use linear or distributed ringing instead of all-ring.
- **Avoid large hunt groups**: Avoid creating hunt groups containing more than 100 members. Distribute the hunt groups over several nodes, if possible.

T1/PRI or E1/PRI Networking

A T1/PRI or E1/PRI connection can be used for connecting two system nodes in a network. Each network span requires two T1M or T1M-2 modules, one for each end of the span. Point-to-point clear channel T1 spans are used to connect the nodes. For each network span, order a clear channel, point-to-point T1 span from the service provider (not PRI service, even though you will be using T1/PRI or E1/PRI). The following figure illustrates a sample T1/PRI connection.

Figure 5: Sample T1/PRI Connection



In a T1/PRI or E1/PRI network that involves multiple CO and T1M or T1M-2 module connections, the line buildout is determined by each connection. In the example shown above, the two T1M or T1M-2 modules connected to the CO receive reference clock from the public network and are slave clocks (Slaves to Public Network Setting in DB Programming). The two modules that are connecting the Mitel systems do not draw reference clock from the public network. T1/PRI #2 on system #1 is the master clock (Master for Private Network setting in DB Programming) and T1/PRI #1 on system #2 is a slave clock (Slave for Private Network setting in DB Programming).

Network installation recommendations include the following:

- Each node *must* have at least one central office trunk. Emergency calls should be programmed to use the local trunk.
- If an external voice processing system is shared by multiple nodes, all of the nodes should have access to the node where the external voice processing system is located, with a minimum number of hops through other nodes. This will have significant impact on the network layout.

The network does not allow one node to use the resources from another node (conference and speakerphone resources, etc.) with the exception of external page ports and message print output.

T1/PRI and E1/PRI Programming

You can also configure T1/E1 PRI networking using the Networking Wizard. This section describes each portion of the networking in DB Programming. For programming information discussed here, refer to the *MiVoice Office 250 Features and Programming Guide*.

A port on the T1M or T1M-2 module can be configured for T1/PRI or E1/PRI networking. The T1/PRI and E1/PRI programming has a field called “ISDN Switch Type” (this field is not present in a module configured for T1 programming). The ISDN switch types provided for networking are Private Networking and IP Private Networking. To program a port on the T1M or T1M-2 module for networking, use the ISDN Switch Type list box to select Private Networking or IP Private Networking.

When the port is programmed for T1/PRI or E1/PRI networking (Private Network switch type), the system automatically creates a node trunk group. (All B-Channels residing on the same port are in the same node trunk group.) The system allows up to 3 node trunk groups per node (one for each T1/PRI or E1/PRI configuration). *MiVoice Office 250 Features and Programming Guide*

The procedure must be completed on each node networking to the other. For example, you want to network node 1 to node 2.*MiVoice Office 250 Features and Programming Guide*

IP Networking

Refer to the *MiVoice Office 250 Features and Programming Guide* for information discussed in this section.

The MiVoice Office 250 can also provide networking using an IP network. You can also configure IP networking using the Networking Wizard (recommended)*MiVoice Office 250 Features and Programming Guide*.

The section here describes each portion of the networking in DB Programming. Installation recommendations include the following:

- Each node *must* have at least one central office trunk. Emergency calls should be programmed to use the local trunk.
- If an external voice processing system is shared by multiple nodes, all of the nodes should have access to the node where the external voice processing system is located, with a minimum number of hops through other nodes. This will have significant impact on the network layout.
- The Base Server provides 6 IP resources for networking. To increase the number of IP networking resources, you must have the IP Networking license. With the IP Networking license in place, you can then decrease the resource allocations for phones, faxing, or networking resources. Refer to the *MiVoice Office 250 Features and Programming Guide* for details. Additionally, faxing consumes two networking resources.

The network does not allow one node to use the resources from another node (DSP resources, etc.) with the exception of external page ports and message print output.

the *MiVoice Office 250 Features and Programming Guide*IP Networking Planning sheets are included at the end of this section for your convenience.

the MiVoice Office 250 Features and Programming Guide**the** MiVoice Office 250 Features and Programming Guidethe MiVoice Office 250 Features and Programming GuideMiVoice Office 250 Features and Programming Guide

Extension Numbers

Each node has an extension and feature code programming. However, the network should have a universal numbering plan so that extension numbers on the various nodes do not overlap and do not conflict with feature codes. That is, when planning the extension numbers for each of the nodes in the network, set aside a block of extension numbers (for phones, hunt groups, Voice Processing applications, etc.) for each node.

Programming Off-Node Extension Numbers

You can give users access to phone, hunt group, Voice Processing application, and page port extensions on other nodes in two ways:

- You can manually program off-node devices (extensions from other nodes) into the database of the node you are programming. To give a node access to an off-node device, program the extension number, username, description, device type (phone, hunt group, Voice Processing application, or page port), and node identification number. The network uses the username and descriptions for directory features only. If you define an extension conflict, DB Programming does not allow you to exit the screen until you correct the conflict.
- You can import and export new extensions as you create them. When you add a new extension, DB Programming automatically makes the new extension visible to the other nodes in the network. Or you can manually export local devices to other nodes (or import devices from other nodes) using the Export/Import option in DB Programming. The network broadcasts the new extension number (along with the username, description, device type, and node identification number) to the other nodes in the network and places it in their off-node device lists. If there is an extension conflict on another node, the node with the conflict ignores the new extension in favor of the previous extension number. Also note that if the network is unable to route the broadcast message to a node, that node will not be able to receive and re-broadcast the new information and it must be entered manually. (If Message Print is enabled, error messages will indicate any unsuccessful broadcasts.)

Each node can have only 10,000 off-node device entries in its database. Once that limit has been reached, new devices received through network broadcasts cannot be added to the node's database. You can make room in the database for new extensions by setting up wildcard extensions as described below.

If you change a device extension, username, or description, the network automatically broadcasts that extension change to the entire network and updates any existing off-node devices on the other nodes. If the network is unable to export or import an extension because there is an active programming session on that node, the node is unable to communicate with its Voice Processing port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. You must manually add or change the new extension in the node database. You may want to check each node to verify that their off-node device lists are programmed properly to allow access between the nodes.

Nodes are assigned extension numbers that users can dial to reach phones which are not programmed as off-node devices on their node. Also, if the same extension number exists on two nodes, the only way a user can call the extension on the other node is to dial a node number before dialing the extension number. For example, if extension 1200 exists on both Node 1 and Node 2, users on Node 1 must first dial the node number for Node 2 to before dialing extension 1200 on

Node 2. (Note that each node trunk group also has an extension, but the number cannot be dialed by users.)

Wildcard Character X in Off-Node Extensions

You can use the wildcard character X in extension numbers to represent “any digit.” This allows you to include a range of extensions as one off-node device entry. Wildcard extensions can only be used for off-node devices.

Wildcard extensions are made up of digits (1–9), followed by wildcard digits X. Valid wildcard extensions include 1XXX (range of 1000–1999), 14XX (range of 1400–1499), 7X (range of 70–79). For example, if 14XX is a wildcard extension, users can then dial a 14XX number, such as 1433, and be connected to the off-node device.

There are some limitations to using wildcard extensions:

- There will be no entry for wildcard extensions in the IC Directory.
- PKM and DSS buttons cannot be programmed for the individual devices the wildcard extension is representing. For example, if the off-node device 4:14XX exists, the user is restricted from programming extensions 1400–1499 as a PKM or DSS/BLF button. As a result of this action, the system displays an error message **INVALID EXTENSION NUMBER** and then re-prompts for another button.
- Voice Processing cannot create extension IDs or associated mailboxes for phones included in wildcard ranges. Each associated phone must have its own off-node device entry.
- Wildcard extensions cannot be used for Call Routing destinations. For example, again using the 4:14XX case, the user is restricted from using unconditional forward feature code (355) and setting the destination to extensions 1400–1499. As a result of this action, the system displays an error message FORWARD TO INVALID DEST and re-prompts for another destination.

Node Programming

Each node is programmed individually and contains its own database. Only one DB Programming session—including administrator database programming—may be active on any single node at a time. However, several nodes can have DB Programming sessions active throughout the system.

Simultaneous sessions on different nodes can cause conflicting database changes, but they also allow the flexibility of easier system administration.

Each node is given a unique identification number. Node numbers 1–63 apply if the 99 Nodes capability is not available. If the main node is a 5000 system running v2.1 or later, nodes may be numbered 1–99. The programmer is responsible for ensuring that each node's identification number is unique and that all of the nodes agree with respect to their node identification numbers.

A node contains a list of node trunk groups used to access other nodes. For each node in the network, you must define the routes to every other node. For example, in a network with four nodes, you would define three routes for each node (one to each of the other three nodes).

Each node contains:

- An optional username
- An optional description
- The identification number of the remote node
- A list of the node trunk groups used to establish calls and communications to the node

IMPORTANT	If desired, you can assign the same node trunk group to multiple nodes. This is a common setup when using IP or frame relay devices, such as the IP Networking Module. However, each node trunk group can have up to 23 trunks (T1/PRI) or 30 trunks (E1/PRI), thus limiting the number of incoming and outgoing calls. Make sure you have enough T1M or T1M-2 modules to support the customer's call load.

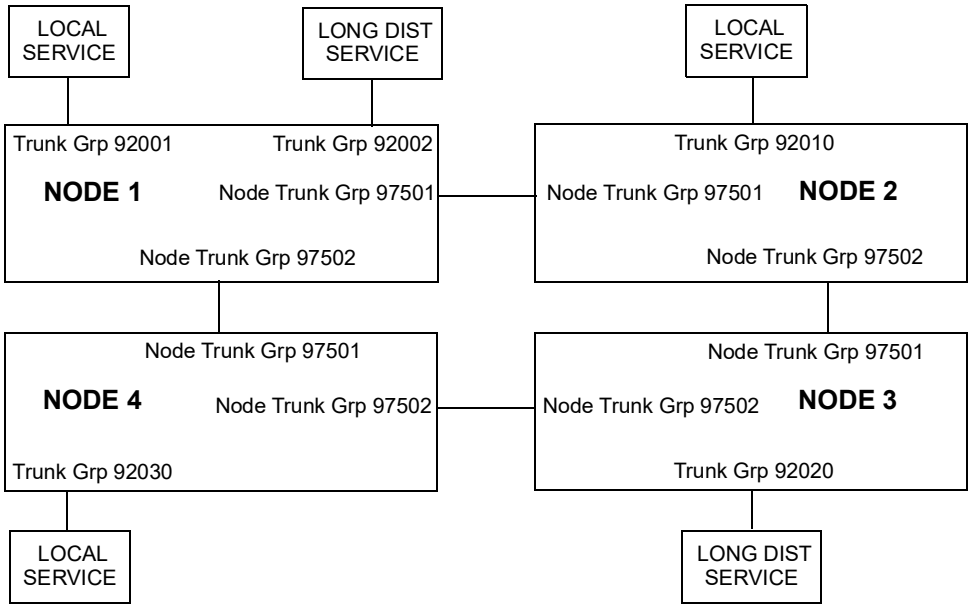
- A search algorithm that determines whether the node trunk groups are accessed in linear or distributed order:
  - **Linear**: The node first attempts to route through the first node trunk group listed in the route. If the node is unable to route through that node trunk group, it attempts to route through the second node trunk group listed in the route. The node continues to attempt to route through subsequent node trunk groups listed in the route until it successfully routes or exhausts all node trunk groups listed in the route.
  - **Distributed**: The node shifts the first node trunk group it attempts to use. For example, if the node routed the previous call or communication through the first node trunk group listed in the route, the node routes the second call or communication through the second node trunk group listed in the route.

If an off-node device is not programmed on the local node for a phone on a remote node, users can still call that device by first dialing the node number for the phone node and then the phone extension number. For example, in 6 on **page 1-401**, a caller on Node 1, calling extension 1100 on Node 2, could dial Node 97002 and then 1100 to reach the phone on Node 2.

Sample Node Programming

The programmer must define the routes to every node in the network to give the system the information necessary to route calls. Each node trunk group, and its B-Channels, in a route group must connect the node to another node in the network. However, that node does not need to be the destination node. For example, consider the following network in **Figure 6**.

Figure 6: Sample Node Programming



Using the map above, a programmer can determine the necessary information about each route to every node in the network, as shown in Table 72, “Sample Network Node Routing Plan,” on **page 401**.

Table 72: Sample Network Node Routing Plan

Node Route Group #	Node #	Description	Node Trunk Groups	Search Type
Programming for Node 1:				
NRG 02 97002	2	To Node 2	97501, 97502	Linear
NRG 03 97003	3	To Node 3	97501, 97502	Distributed
NRG 04 97004	4	To Node 4	97502, 97501	Linear
Programming for Node 2:				
NRG 01 97001	1	To Node 1	97501, 97502	Linear
NRG 03 97003	3	To Node 3	97502, 97501	Linear
NRG 04 97004	4	To Node 4	97501, 97502	Distributed
Programming for Node 3:				
NRG 01 97001	1	To Node 1	97501, 97502	Distributed
NRG 02 97002	2	To Node 2	97501, 97502	Linear

(Sheet 1 of 2)

Table 72: Sample Network Node Routing Plan (Continued)

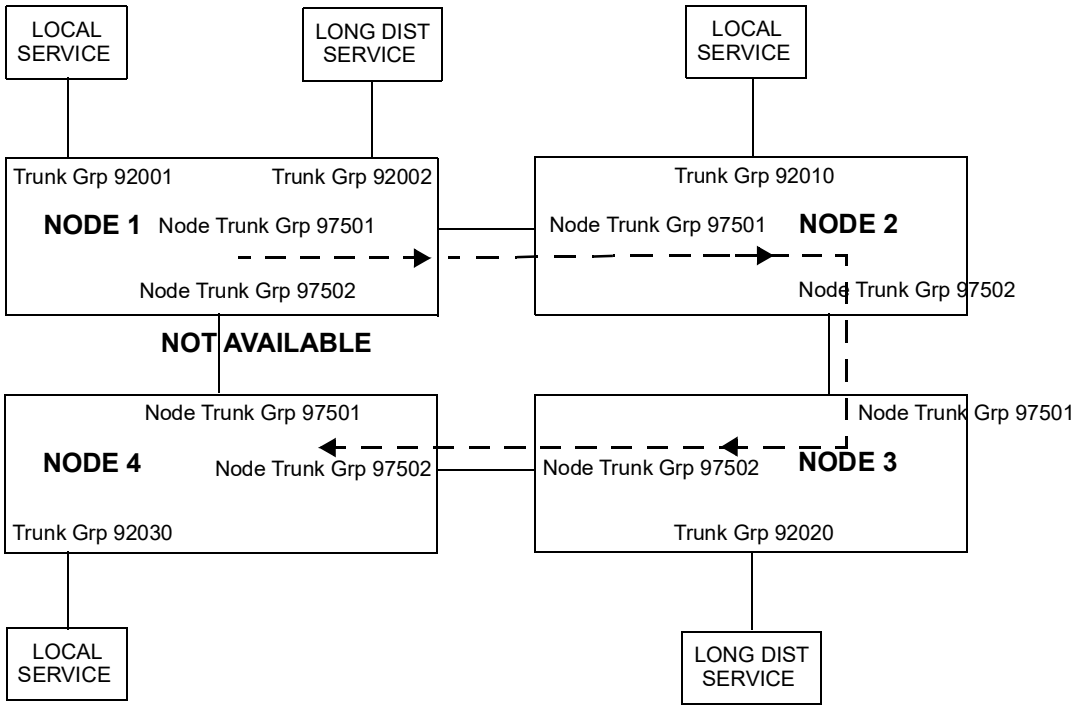
Node Route Group #	Node #	Description	Node Trunk Groups	Search Type
NRG 04 97004	4	To Node 4	97502, 97501	Linear
Programming for Node 4:				
NRG 01 97001	1	To Node 1	97501, 97502	Linear
NRG 02 97002	2	To Node 2	97501, 97502	Distributed
NRG 03 97003	3	To Node 3	97502, 97501	Linear

(Sheet 2 of 2)

To prevent circular or excessive routes, the system limits the number of “hops.” A call or communication can take up to 10 hops to other nodes. However, 2-3 hops maximum is recommended. Using the example shown in 6 on **page 1-401**, 7 on **page 1-403** shows a call that makes three hops through intermediate nodes to reach a destination:

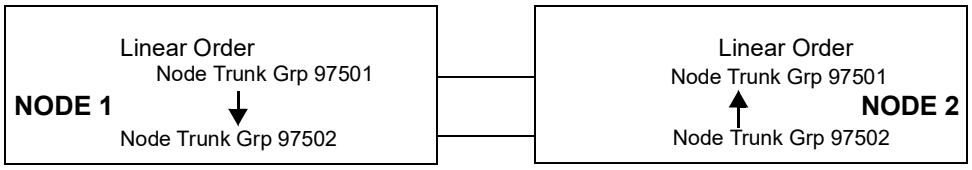
1. A call is placed from Node 1 to Node 4.
2. Node 1 attempts to route through Node Trunk Group 97502, but that route is not available. Node 1 then routes the call using Node Trunk Group 97501 and uses Nodes 2 and 3 as intermediate nodes.
3. Because the network has already routed the call through Node 1, Node 2 will not use Node Trunk Group 97501 and routes the call through its Node Trunk Group 97502.
4. Node 3 routes the call through its Node Trunk Group 97502 to Node 4. It will not use 97501 because the call came from Node 2.

Figure 7: Example Call Path Through Nodes



Determining in what order to list the Node Trunk Groups in a Node and when to use Linear or Distributed search type depends on the system configuration and traffic.

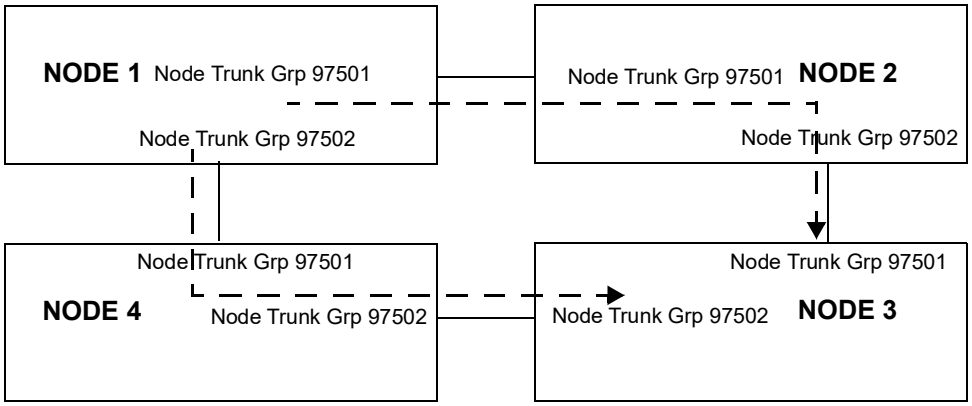
For example, **Linear** search type is the preferred method for avoiding glare between two nodes that have two or more connections. Place the Node Trunk Groups in opposite order in the Nodes on each side of the connection and use Linear search type, as shown below.



Node Route Group #	Node #	Description	Node Trunk Groups	Search Type
Programming for Node 1:				
NRG 02 97002	2	To Node 2	97501, 97502	Linear
Programming for Node 2:				
NRG 01 97001	1	To Node 1	97502, 97501	Linear

In the above example Node 1 uses Linear searches to send calls first through Node Trunk Group 97501 and then through Node Trunk group 97502. Node 2 reverses the order and begins searching with Node Trunk Group 97502. With calls beginning at opposite ends of the trunk lists, it is less likely that both nodes will try to seize the same trunk and cause glare.

Distributed search type is best for balancing traffic when there are multiple-hop paths between nodes. In the example shown below, the path to Node 3 from Node 1 has two hops using either Node Trunk Group. Therefore, the route from Node 1 to Node 3 should be Distributed to balance the call traffic.



Node Route Group #	Node #	Description	Node Trunk Groups	Search Type
Programming for Node 1:				
NRG 03 97003	3	To Node 3	97501, 97502	Distributed

Node Programming Procedures

Program the network in the following order:

1. Local Node Number
2. Nodes
3. T1M or T1M-2 modules set up for T1/PRI or E1/PRI
4. Node Trunk Groups
5. Node Trunk Lists
6. Off-Node Devices

Local Node Number

To program a node local node number and description:

1. Make sure you have the PRI software license.
2. In System\Devices and Feature Codes\Nodes, program the local node number and the local node description. This causes the system to reset, and you will have to re-establish the DB Programming connection.

Nodes

To program each of the nodes:

1. In System\Devices and Feature Codes\Nodes, create the Node.
2. Set the remote node number to the destination node.
3. Name the node (for example To Node X).
4. Repeat for all nodes.

T1M or T1M-2 Modules

To program the T1M or T1M-2 modules:

1. Program the module bays and corresponding ports, then program B-channels and extension numbers for the circuits.
2. Select the port to program, then set the ISDN Switch Type to **Private Network**. This will automatically place the trunks on this port in a node trunk group.
3. Set the Remote Node Number to the node where the T1 span will be connected. (This span can also be used to route calls to other nodes, through the remote node, if the remote node is properly programmed.)

Node Trunk/IP Connection Groups

To program the node trunk/IP connection groups:

1. In System\Devices and Feature Codes\Node Trunk/IP Connection Groups, select the node trunk group.

2. Name the node trunk group, set outgoing access assignments.
3. Repeat for the remaining node trunk/IP connection groups.

Node Trunk Lists

To add the node trunk groups to the node list:

1. In System\Devices and Feature Codes\Nodes, select the remote node.
2. Select Node Trunk/IP Connection Groups and add the node trunk groups.
3. Program the Search Algorithm.
4. Repeat for each node.

Off-Node Devices

There are two methods for programming off-node devices, as described on **page 398**. The automatic broadcast method works only for newly created phones. Any extensions that existed before the nodes began communicating must be exported (or imported) to other nodes using the Export/Import button in DB Programming or programmed as “off-node devices” manually at each of the other nodes.

Network Operation of System Features

When a MiVoice Office 250 is programmed for use in a network the features listed here are affected. Refer to the features information in the *MiVoice Office 250 Features and Programming Guide* for details.

- **Administrator Feature Codes**: *MiVoice Office 250 Features and Programming Guide*
  - **Clear Network Alarm:** Administrators who receive network-wide alarms can clear them. The default feature code for the clear network alarm feature is 9851. See **page 412** for more information on network alarmsthe *MiVoice Office 250 Features and Programming Guide*.
  - **Enable Network Night or Day Mode:** Administrators can place any or all remote nodes into night mode or day mode. The default feature code for Enable Network Night is 9861. The default feature code for Enable Network Day is 9862. The Night Ring On/Off feature code (9860) affects only the node on which the administrator resides. the *MiVoice Office 250 Features and Programming Guide*
  - **Set Network Date And Time:** Administrators can synchronize the time of day, date, and day of week across the network. The default feature code for the synchronize network time of day feature is 9810. Note that administrators can still set the time of day, date, and day of week on their local node. the *MiVoice Office 250 Features and Programming Guide*
  - **Synchronize Network Time:** Administrators can synchronize the minutes past the hour across the network without changing the hour. This is useful when the nodes are in different time zones. The default feature code for the synchronize network minutes past the hour is 9811. If the node time is off by more than 30 minutes, synchronizing the minutes may cause the hour to change. Also, network time is automatically synchronized every day at 12:30 AM (00:30), using the time setting on the node with the lowest number. the *MiVoice Office 250 Features and Programming Guide*
  - **Network Freeze/Unfreeze Zone:** Administrators can perform a history freeze or unfreeze on any freeze zone. The default feature code for Network Freeze is 9939 and Network Unfreeze is 9989. the *MiVoice Office 250 Features and Programming Guide*
- **Administrator DB Programming**: The administrator can change extensions and usernames of only those users residing on the same node as the administrator. The network prohibits the administrator from programming an extension conflict. The network automatically broadcasts changes to extensions and/or usernames made by an administrator to the other nodes in the network. the *MiVoice Office 250 Features and Programming Guide*
- **Attendant Phones**: An attendant does not need to reside on the same node as the phones it serves. the *MiVoice Office 250 Features and Programming Guide*
- **Node Attendants**: There can be a primary attendant for each node. *MiVoice Office 250 Features and Programming Guide*
- **Network Primary Attendant**: There can be a network primary attendant programmed on each node. (There is usually one Network Primary Attendant shared by all of the nodes.) When the network needs to direct a call to an attendant, it attempts to direct the call to the network primary attendant first. If the network primary attendant is unavailable because the network is unable to route the call, it directs the call to the attendant on the node where it originated. the *MiVoice Office 250 Features and Programming Guide*

- **Extension Numbers**: When an extension number is changed or deleted, the network broadcasts the extension change to the other nodes in the network, and the network updates all off-node device lists appropriately. If the network is unable to export or import an extension because there is an active programming session on that node, the node is unable to communicate with its Voice Processing port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. You must manually change or delete the extension from the node database.
- **Extension Lists:** Extension lists are limited to the phones or trunks on a single node. *MiVoice Office 250 Features and Programming Guide*
- **Hunt Groups:** UCD hunt group members can be off-node devices. ACD hunt group members can also span nodes. Supervisors, however, cannot span nodes. All versions support supervisors, recall destinations, announcement phones, and overflow phones on other nodes. the *MiVoice Office 250 Features and Programming Guide*
- **Phone Monitor:** Because supervisors must reside on the same node as the hunt group, phone monitoring can not occur across nodes. the *MiVoice Office 250 Features and Programming Guide*
- **Trunk Group**s**:** the *MiVoice Office 250 Features and Programming Guide*
  - **Trunk Lists:** All trunks in a trunk group must reside on the same node as the trunk group.
  - **Outgoing Access:** Because the network does not allow users to directly access trunks on other nodes, each trunk group's Outgoing Access must contain only phones on the local node. Users must use ARS to access trunks on other nodes. Once an ARS call leaves the originating node, the network ignores outgoing access restrictions.
  - **Ring-In Assignment:** A trunk can ring in to a phone or Voice Processing application on another node.
  - **Allowed-Answer Assignment:** Phones cannot have allowed-answer assignment for trunk groups on other nodes.
- **Caller ID, DNIS, and ANI**: The network retains trunk and outside party information and passes that information along when a call is routed from node to node. the *MiVoice Office 250 Features and Programming Guide*
- **Call Routing Tables**: Call routing tables can be used on a network. the *MiVoice Office 250 Features and Programming Guide*
  - You cannot route a call from one Call Routing Table to a Call Routing Table on another node. When a Call Routing Table sends a call to another node, it cannot be routed to any additional nodes.
  - Trunks that ring in to call routing tables can only ring in to tables that are located on their local node. They cannot ring in to a call routing table on another node.
  - The network attempts to optimize routes so that calls do not route through a node more than one time (looping). However, the network does not have the ability to dynamically reroute. That is, if a node trunk group that was previously unavailable becomes available after the network has already chosen another route, the network does not reroute using that node trunk group.
  - The network will try to optimize routes when calls have been transferred or reverse transferred across several nodes (possibly forming loops).

- **Automatic Route Selection (ARS)**: Each node has its own ARS programming. All trunk groups and nodes in a facility group must reside on the same node as the facility group. You cannot place a node ***trunk*** group into a facility group. (See also Outgoing Access on this page.) Using ARS is the only way a user can access trunks on other nodes. If the system routes an outgoing call to another node using an intermediary node, the intermediary node's ARS cannot use the nodes in its facility group to move the outgoing call to another node. This eliminates the system from getting into an infinite loop searching for a node to route the outgoing call. the *MiVoice Office 250 Features and Programming Guide*
- **ARS Toll Restriction**: When determining toll restriction for an ARS outgoing call, the network only checks the phone toll restriction for the node on which the phone resides. The system does not check the trunk class of service for ARS calls.
- **Emergency Calls**: Emergency calls, by default, use the first local trunk group and will not be sent using node trunk groups on other nodes. However, when ARS is used to place an emergency call, Route Group 1 is used even if it contains nodes. This means that the network ***can*** access a trunk on a node other than the user's node if the user accesses ARS and dials the emergency number. ***Mitel strongly recommends that local trunks be installed and used for emergency number trunk access and that nodes not be used in Route Group 1.*** When a user places an emergency call, every administrator in the network receives an emergency alarm. **the** *MiVoice Office 250 Features and Programming Guide*
- **Direct Inward System Access (DISA)**: DISA callers can select off-node phones, hunt groups, and Voice Processing applications.
- **Keymaps**: Each node contains its own keymap programming.
- **PKM or DSS/BLF Keys**: You can program a PKM or DSS button for an extension that resides on another node. The buttons will show lamp status for phones on the other nodes.
- **Secondary Extension Keys**: The primary phone and the secondary phone must reside on the same node. the *MiVoice Office 250 Features and Programming Guide*A secondary extension cannot be a phantom extension.
- **Music-On-Hold and Background Music**: Each node has a music source. If a caller on Node 1 is holding for a user on Node 2, the caller hears the music on node 2. Note that if a caller is holding for a user on Node 1 and the call moves to Node 2 (due to a transfer, forward, or recall), the user will stop hearing the music on Node 1 and will begin hearing the music on Node 2. **the** *MiVoice Office 250 Features and Programming Guide*The system can be programmed to determine the music source a caller hears based on the device for which the caller is waiting.
- **Handsfree Calls**: The network allows handsfree intercom calls when calling from one node to another.
- **Camp On**: A user can camp on to busy resources on another node. When the system has several routing options for a call, it tries to use each route, in order, until it finds a free route. If it is unable to find a free route, the system tries once again to use the first route. If the first route is still busy, the system camps on to the first route. the *MiVoice Office 250 Features and Programming Guide*
- **Queue**: A user cannot queue onto busy resources on another node because Queue is not supported on a network. the *MiVoice Office 250 Features and Programming Guide*
- **Inter-Station Messages**: The network allows users to leave messages for, and respond to messages from, users on a different node. The phone message center or alternate message

source does not need to be on the same node as the phone. It can be a phone, hunt group, Voice Processing application, or any off-node device.

- **Off-Hook Voice Announce (OHVA)**: The network allows intercom callers to establish OHVA calls to phones on other nodes. the *MiVoice Office 250 Features and Programming Guide*OHVA is not functional on IP phones.
- **Account Codes**: The network validates account codes against the account code table on the same node as the caller. The account code follows the call as it moves from node to node and appears on every SMDR record associated with the call. the *MiVoice Office 250 Features and Programming Guide*However, if an optional account code is entered during the call, that account code will be used in the SMDR report.
- **System Hold**: When a user puts a trunk on system hold, only that user and users on the same node as the trunk can access the call. the *MiVoice Office 250 Features and Programming Guide*
- **Transfer Recall Destination**: The phone transfer recall destination can be an off-node device. the *MiVoice Office 250 Features and Programming Guide*
- **Call Screening**: In a network setting, the Voice Processing can provide call screening for a destination extension on another node. However, the node where the Voice Processing is connected must have an off-node device programmed for the destination extension and access to the remote node. the *MiVoice Office 250 Features and Programming Guide*
- **Conference Calls**: A user can build a conference with party members (phones and/or trunks) on other nodes. The conference circuits used to build the conference will be allocated from the originating party's node. Any party, not just the originating party, can add a party member to the conference. the *MiVoice Office 250 Features and Programming Guide*
  - If the party who adds the new member is on the originating party's node, the party will be added to the conference circuit already allocated. This conference is still restricted to the four party limit.
  - If the party who adds the new member is not on the originating party's node, a new conference circuit is allocated from the second node. Because this allows more than four parties in a conference, users should be warned that voice levels can be degraded if more than four parties are on a conference.
- **Reverse Transfer**: A user can reverse transfer a call from a phone or a hunt group on another node. the *MiVoice Office 250 Features and Programming Guide*
- **Record-A-Call**: The Record-A-Call destination does not have to be on the same node as the phone, but the EM unit must have a mailbox for that phone. the *MiVoice Office 250 Features and Programming Guide*
- **Agent Help**: The Agent Help Extension does not need to reside on the same node as the user requesting Agent Help. the *MiVoice Office 250 Features and Programming Guide*
- **System Forwarding**: System forward points can reside on other nodes. the *MiVoice Office 250 Features and Programming Guide*
- **Call Forwarding**: The network allows the user to program a manual forward destination to phones on another node. the *MiVoice Office 250 Features and Programming Guide*
- **System Speed Dial**: Each node in a network has its own system speed dial numbers. System speed dial numbers can be used only on the node on where they are programmed. the *MiVoice Office 250 Features and Programming Guide*

- **Directories**: The network displays intercom directory names and extension numbers for all devices the user can access directly. Devices that are represented by wildcard extensions will ***not*** show up in the intercom directory. the *MiVoice Office 250 Features and Programming Guide*
- **House Phone**: The House Phone can be programmed to dial the extension number of an off-node device. the *MiVoice Office 250 Features and Programming Guide*
- **Paging**: All phones and trunks within a page zone must reside on the same node as the page zone. However, a page zone can contain external page ports on other nodes and off-node page zones on the local node. the *MiVoice Office 250 Features and Programming Guide*
- **Do-Not-Disturb**: Do-Not-Disturb works as follows in a network setting. the *MiVoice Office 250 Features and Programming Guide*
  - Each node has its own list of Do-Not-Disturb messages that can be used only on that node.
  - An intercom caller will see another user's Do-Not-Disturb display when calling across nodes.
  - The network allows DND override across nodes.
- **Remote Feature Programming**: Remote Feature Programming is used for setting a phone's Do-Not-Disturb and call forwarding modes from another phone or through DISA. Remote programming can be performed on one node for any phone that is programmed as an off-node device on that node. You must go through DISA for remote programming of any phones that are not programmed as off-node devices. the *MiVoice Office 250 Features and Programming Guide*
- **Reminder Messages**: Each node has its own list of reminder messages that can be used only on that node. the *MiVoice Office 250 Features and Programming Guide*
- **Day and Night Modes**: The network determines the day/night mode status of a call based on the day/night mode status of the node where the trunk resides. the *MiVoice Office 250 Features and Programming Guide*
- **Call Cost**: The call cost shown on the display and SMDR output use the factors and rates for the node on which the trunk resides. In other words, if a phone on Node A dials what is considered a long-distance number on node A, but the call is routed to Node B where the number is considered local, the phone will use the local call cost rate from Node B. The call rate used for calls between nodes will be based on the Network call rate. the *MiVoice Office 250 Features and Programming Guide*
- **SMDR**: Each node has its own SMDR programming, SMDR output port, and SMDR output port backup. the *MiVoice Office 250 Features and Programming Guide*You can enable or disable network call records on each node. When network calls are enabled, the following applies:
  - The system generates SMDR records for outgoing calls on the node where each trunk used resides. For example, if a caller on Node A places a call using a trunk group on Node B, the SMDR report for Node A will show the outgoing call to Node B and the SMDR reports on Node B will show the incoming call from Node A and the outgoing call on the trunk group.
  - The call record type for network SMDR records is NET.
  - If the node SMDR output port is a node, the network sends SMDR records to the specified node.
  - You cannot select a node as the SMDR output port backup.
  - If SMDR output programming forms a loop, the system will send the SMDR output to the node backup SMDR port. For example, if the SMDR port on Node 1 routes to Node 2 and the SMDR port on Node 2 routes to Node 1, the configuration will cause an infinite loop of

SMDR routing. SMDR reports for Node 1 would be printed to the backup port on Node2 and vice versa.

- **Voice Mail Messages**: To correct for time zone differences when message notification is sent from EM on one node to phones on other nodes, there is a Time Zone setting in each mailbox. If set properly, the voice mail message envelope will indicate the time the message was received using the mailbox's programmed time zone.
- **SMDR Buffering**: The optional EM will buffer SMDR records that are sent from other nodes in the network.
- **Message Print**: Each node has its own message print programming, message print output port, and message print output port backup. There should be a message print terminal at each node to monitor node and network performance and aid in troubleshooting. the *MiVoice Office 250 Features and Programming Guide*
  - If the node Message Print output port is a node, the network sends Message Print records to the specified node.
  - You cannot select a node as the message print output port backup.
  - If Message Print output programming forms a loop, the system will send the output to the node backup Message Print port. For example, if the Message Print port on Node 1 routes to Node 2 and the Message Print port on Node 2 routes to Node 1, the configuration will cause an infinite loop. Message Print reports for Node 1 would be printed to the backup port on Node2 and vice versa.
  - A two digit node number is inserted in front of the sequence number to indicate the node where the message originated. The following is an example of the new message print string format for a message that originated on node 3:

    -**03**:001- 15:30 03-20 M6009 WRN EG CP Corrupted Queue Was Cleared: HISTORY QUEUE
  - Message Print will ***not*** route diagnostic messages that apply to another node, except during manual Message Print requests.
  - Message Print will not send messages during a startup or reset. The system will wait one minute before starting to send messages. Startup messages will only be printed locally.
  - Messages may not be printed in time sequence order on remote nodes. The order of an incoming message print depends on how the message print was routed to the destination node. Therefore, a message print could possibly arrive at the destination message print node out of order.
- **System OAI**: System OAI Applications that support networking can be used in conjunction with the CT Gateway/Transceiver. For detailed information about the Gateway/Transceiver, see **page 328**. The Gateway allows a single System OAI application to support a multi-node network. Each application will require a separate Gateway.

  Some System OAI applications that do not support networking will not operate properly on a MItel system that is part of a network, even with a Gateway. This is because those applications are not designed to track calls that cross over from node to node within a network. Those System OAI applications can only be used on stand-alone Mitel systems that are not part of a network.

Network-Wide Alarms

To allow one administrator to monitor multiple nodes, the system provides both system alarms and network-wide alarms:

- **Network-Wide Alarms:** When an event occurs that generates a network-wide alarm, the alarm is broadcast to every node in the system.
- **System Alarms**: System alarms appear only on the node on which the alarm was generated.

Sending and Receiving Network Alarms

Each node has two system flags that determine whether the node broadcasts or receives network-wide alarms.

- The **Send Network Alarms** flag determines whether a node will broadcast alarms that occur on that node to the rest of the network.
- The **Receive Network Alarms** flag determines whether the node will receive and display alarms sent by other nodes in the network.

To differentiate between network-wide and local alarms, network-wide alarms appear on administrator phones preceded by “NET ALARM” and local system alarms are preceded with “SYS ALARM.”

On remote nodes, network-wide alarms will indicate the name of the node on which the alarm occurred. The node name is obtained from the node username.

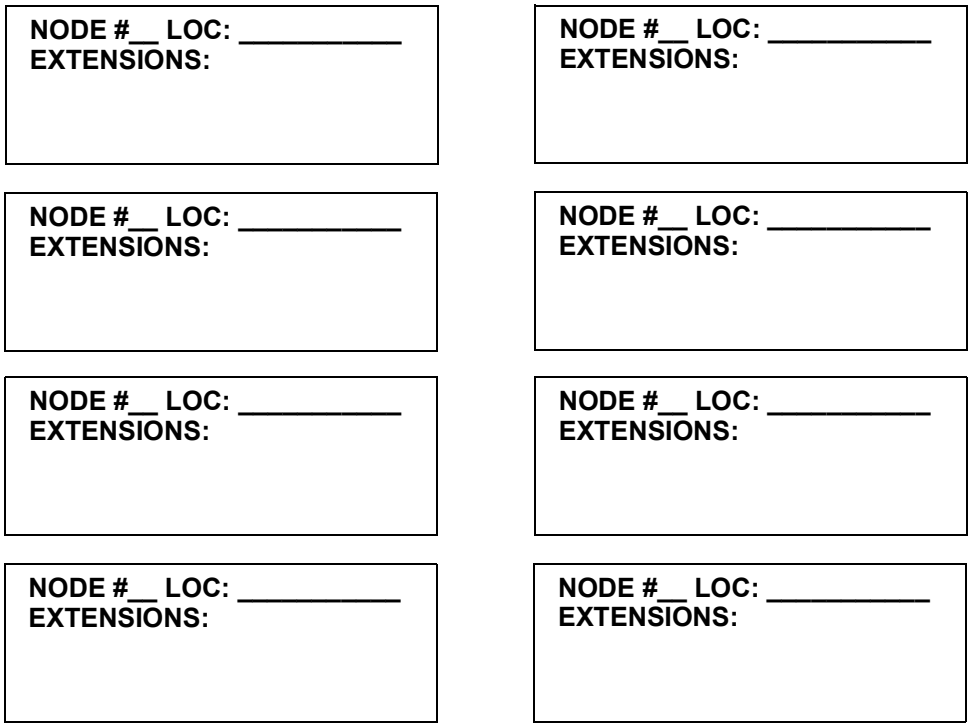
Clearing Network Alarms

An administrator can clear a network-wide alarm on the local node only or on every node in the network.

- **Clear Network Alarm (9851):** This feature code clears network-wide alarms on every node in the network, but will not affect system alarms. The Clear Network Alarm feature code may be entered on any node in the network, but the Send Network Alarms flag must be set for the administrator to clear alarms on other nodes in the network.
- **Clear System Alarm (9850):** This feature code clears all local and network-wide system alarm displays on your node.

Network-wide alarms will override system alarms on an administrator’s display phone.

Figure 8: Network Planning Sheet



Use the drawing above, or one similar to it, to map your network. For a sample network map, see **page 401**.

Add information to the drawing base on answers to the following questions:

- How many nodes are in the network? Draw additional Nodes if necessary.
- How are they numbered? Label each Node with its number.
- Where are they located? Label each Node with its location.
- How will the network be configured? Draw lines showing the connections between the Nodes.
- From where will calls ring in to EM and where will they be routed? Show the ring-in route on the map.
- Will call routing tables and/or ARS route calls to other nodes? If so, show these routes on the map.
- Where will the SMDR and Message Print output port(s) be located? If they cross between Nodes, show them on the map.
- Does each node have a local trunk group for emergency calls? Draw and label the local trunk groups for each node.
- Which nodes will have EM? Draw the unit(s) on the appropriate Node(s). Record the EM numbering plan in the worksheet on the next page.
- What is the numbering plan for the extensions, feature codes, hunt groups, node trunk groups, etc? Label your map and/or use the Numbering Plan Worksheet on the next page for recording this information.

Figure 9: Network Numbering Plan Worksheet

Determine the following numbers for the network:

EXTENSION NUMBERS:

Network Primary Attendant:	
Node 1 Range Of Extensions:	
Node 2 Range Of Extensions:	
Node 3 Range Of Extensions:	
Node 4 Range Of Extensions:	
Node 5 Range Of Extensions:	
Node 6 Range Of Extensions:	
Node 7 Range Of Extensions:	
Node 8 Range Of Extensions:	

Determine the following numbers for each Node:

NODE # _____ LOCATION _____

Local Extension Numbers:	
Local Trunk Group Numbers:	
Local Attendant:	
Off-Node Devices: (Include Voice Processing applications if they are not on this node)	
Off-Node Wildcard Numbers:	
Node Trunk Group Numbers:	

Figure 10: IP Networking Planning Worksheet (example)

The following is an example of a completed IP Private Networking Planning sheet. A blank sheet is provided on the following page for your use.

	Node 1	Node 2	Node 3	Node 4
Extensions	1100-1199	1200-1299	1300-1399	1400-1499
Area Code	602	514	404	713
Nodes (Node Route Group)	97001	97002	97003	97004
IP Connection Group	P6001	P6002	P6003	P6004
Node IP Connection Group[a]	P8000	P8001	P8002	P8003

	Node 1	Node 2	Node 3	Node 4
IPRA (Networking)	192.168.0.41	192.168.0.42	192.168.0.43	192.168.0.44
IPRA (Station)	192.168.0.51	192.168.0.52	192.168.0.53	192.168.0.54
MGCP	192.168.0.61	192.168.0.62	192.168.0.63	192.168.0.64
IPPhone Plus	192.168.0.71	192.168.0.72	192.168.0.73	192.168.0.74
Subnet	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0
Default Gateway	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200

a. The system automatically assigns the Node Trunk/IP Connection Group extension numbers.

	Node 5	Node 6	Node 7	Node 8	Node 9
Extensions	1500-1599	1600-1699	1700-1799	1800-1899	1900-1999
Area Code	216	914	802	440	203
Nodes (Node Route Group)	97005	97006	97007	97008	97009
IP Connection Group	P6005	P6006	P6007	P6008	P6009
Node IP Connection Group[a]		P8004	P8005	P8006	P8007
IPRA (Networking)	192.168.0.45	192.168.0.46	192.168.0.47	192.168.0.48	192.168.0.49
IPRA (Station)	192.168.0.55	192.168.0.56	192.168.0.57	192.168.0.58	192.168.0.59
MGCP	192.168.0.65	192.168.0.66	192.168.0.67	192.168.0.68	192.168.0.69
IPPhone Plus	192.168.0.75	192.168.0.76	192.168.0.77	192.168.0.78	192.168.0.79
Subnet	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0	255.255.252.0
Default Gateway	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200	192.168.0.200

a. The system automatically assigns the Node Trunk/IP Connection Group extension numbers.

Figure 11: IP Networking Planning Worksheet (blank)

	Node 1	Node 2	Node 3	Node 4
Extensions				
Area Code				
Nodes (Node Route Group)				
IP Connection Group				
Node IP Connection Group[a]				
IPRA (Networking)				
IPRA (Station)				
MGCP				
IPPhone Plus				
Subnet				
Default Gateway				

a. The system automatically assigns the Node Trunk/IP Connection Group extension numbers.

	Node 5	Node 6	Node 7	Node 8	Node 9
Extensions					
Area Code					

	Node 5	Node 6	Node 7	Node 8	Node 9
Nodes (Node Route Group)					
IP Connection Group					
Node IP Connection Group[a]					
IPRA (Networking)					
IPRA (Station)					
MGCP					
IPPhone Plus					
Subnet					
Default Gateway					

a. The system automatically assigns the Node Trunk/IP Connection Group extension numbers.

Appendix B

Network IP Topology

Introduction

The MiVoice Office 250utilizes network communications for several purposes including administration, IP phones, IP trunks, and private networking. The MiVoice Office 250 is limited to a single network connection shared among the different functions. This appendix discusses the security and associated network topology issues for the MiVoice Office 250. Network suitability and associated performance for VoIP traffic is beyond the scope of this appendix.

Terminology

The following terminology appears in this appendix:

- **ACL** (Access Control List): An ACL is a list of filters (restrictions) applied to communications passing through a router. Generally, each entry specifies a source, a destination, and a port. The source and destination can be individual IP addresses or subnets.
- **DMZ** (Demilitarized Zone): A DMZ is a subnet that is “between” an organization’s private network and the Internet. Communication between nodes in a DMZ, the Internet, and the internal network is controlled by a firewall. A DMZ provides some firewall protection for nodes that need to be available to the Internet, while also protecting the internal network from the DMZ nodes.
- **Audio and Web Conferencing**: AWC is a Mitel product that provides IP-based conferencing capabilities to Mitel systems.
- **Firewall**: A firewall is a network device designed to provide limited connectivity between an organization’s network and the Internet. A firewall generally prevents access from the Internet to an inside network while allowing access from the inside network to the Internet. Some firewalls also provide the capability for a special network, called a DMZ, that is placed between the Internet and the internal network.
- **Private Networking**: This is the protocol used by the MiVoice Office 250 to provide networking functionality.
- **IPRA** (Internet Protocol Resource Application): This is the function on the MiVoice Office 250 that provides IP connectivity to IP phones, trunk gateways, and Private Networking.
- **MGCP** (Media Gateway Control Protocol): MGCP is a standard protocol for commanding VoIP media gateways.
- **NAT** (Network Address Translation): A NAT is an Internet standard that enables a local-area network (LAN) to map between one set of IP addresses for internal traffic and a second set of addresses for external traffic.
- **NAT IP Address**: This is the public IP address that is statically mapped to the MiVoice Office 250 system IP address.
- **Native IP Address**: This is the IP address of the MiVoice Office 250 network interface. This is the MiVoice Office 250 real IP address.
- **OAI**: (Open Application Interface): System OAI is a system-level communications protocol that allows command-oriented tools to be developed for the phone system. System OAI is available in two levels: Events and Third Party Call Control. System OAI Events provides call status information to an external device. Its primary use is for automatic call distribution products such as the MiContact Center Office. System OAI Third Party Call Control is a TSAPI protocol that

allows system command-oriented tools to be used on the phone system. Contact Mitel for more information regarding products and development materials for these features.

- **Port**: In Internet Protocol terminology, a port is a logical subaddress of an IP address. The port usually indicates the application layer protocol being used (for example, TCP port 80 indicates http).
- **Private**: In networking terminology, private refers to the portion of a network within an organization. This private network is considered trusted. The private network is separated from the public network (the Internet) by a firewall.
- **Public**: In networking terminology, public refers to the Internet. This network is considered untrusted because there are few controls about what it contains. The private network is separated from the public network (the Internet) by a firewall.
- **Router**: A router is a layer-3 device (OSI Model) that provides intelligent packet forwarding according to rules that describe the subnet topology. Routers usually make routing decisions based on IP address. Modern routers often contain some firewall functionality.
- **SIP** (Session Initiation Protocol): This protocol is an IETF standard for Internet telephony, conferencing, etc. It is a peer-to-peer text-based protocol focused on signaling setup and teardown. SIP is not a media protocol.
- **SPI** (Stateful Packet Inspection): SPI is provided by firewalls that allow IP communication to take place if it is initiated from the trusted network – the firewall keeps track of which sessions (conversations) are active (state).
- **Subnet**: A subnet is a grouping of devices that share common routing information.
- **TFTP** (Trivial File Transfer Protocol): TFTP is a simplified Internet standard for file transfers used primarily for loading configuration and software updates to IP devices. Unlike the more common FTP, TFTP provides no security mechanisms.
- **VPN** (Virtual Private Network): A VPN is a network that uses a public medium, typically the Internet, to provide connectivity to a logically private network. VPNs use encrypted tunnels to carry the private traffic in a secure manner.

Network Address Translation Overview

Network Address Translation (NAT) is an Internet standard[1] that enables a local-area network (LAN) to use one set of IP addresses for internal traffic and a second set of addresses for external traffic. Generally, the NAT function is provided by a router or firewall.

The main purpose of NAT is to allow an organization to use a pool of (private) IP addresses that is separate from (public) Internet IP addresses. This alleviates a shortage of public IP addresses. Although NAT can provide a limited amount of security, it is rarely used standalone for security purposes[2].

To facilitate NAT, the Internet Assigned Numbers Authority (IANA) has designated certain IP addresses to be private[3]. This designation means that these IP addresses are not valid (routable) on the public Internet. This allows organizations to safely use these addresses within their networks. The designated private addresses are:

- 10.0.0.0 - 10.255.255.255 (One Class A Subnet)
- 172.16.0.0 - 172.31.255.255 (16 Class B Subnets)
- 192.168.0.0 - 192.168.255.255 (256 Class C Subnets)

NAT operates by dynamically associating each internal private IP address with an external public IP address (and port). The NAT box (router or firewall) keeps track of the association between internal and external addresses and re-writes the IP packet header addresses as necessary. The association between internal and external IP addresses is generally short-lived based on activity.

Although NAT has been widely used throughout the Internet, some protocols do not work well with NAT. As described above, NAT translates the IP addresses in only the IP packet headers. The root of the problem is that some protocols carry IP addresses in the IP packet payload. As a result, private IP addresses are sometimes communicated out to the public Internet. By design, these private IP addresses are not accessible.

It is also possible to configure persistent or static NAT assignments in which a specific outside address is associated with a specific inside address. Although static NAT essentially allows an inside device to be accessible from the outside, it is still NAT and therefore problematic for some protocols when IP addresses are carried in the IP packet payload.

Some Internet applications (for example, IP telephony) do not allow use of NAT. Some firewall vendors offer capability to “fix” NAT problems for specific protocols (for example, SIP).

Because NAT takes place where a private network connects to a public network, there are often two NAT operations taking place – one at each end of the communication. The NAT operations are the same at the two ends, but sometimes the impact on network protocols is different. For the purposes of this discussion, the NAT that takes place between the server (for example, a MiVoice Office 250) and the public network is referred to as near-end NAT. The NAT that takes place between IP phones and the public network (as in a home network) is referred to as far-end NAT.

1. Defined in RFC3022.
2. This is because the public address/port combinations can be easily guessed. Even with dynamic NAT, the address associations are open for long periods of (computer) time during which the internal computer is vulnerable to attack.
3. Defined in RFC1918.

There is a variation of NAT called PAT (Port Address Translation) in which the trusted IP address and port are translated dynamically to an outside IP address and (different) port. For the purposes of this appendix, the term NAT includes true PAT as well as true NAT.

Communication Types

The following communication types are discussed in this appendix.

- **Administration** below
- **IP Phones and Trunks** below
- **Private Networking** below
- Computer Telephony Integration (CTI) on **page 423**
- Audio and Web Conferencing on **page 423**
- System Manager on **page 423**

Administration

The MiVoice Office 250 provides administrative functions through its network connection. These functions include:

- DB Programming
- System Manager
- Web-based diagnostics
- Diagnostics Monitor
- TFTP: Trivial File Transfer Protocol
- SSH/Telnet/USB Console

IP Phones and Trunks

The MiVoice Office 250 communicates to phones through the internet protocol (IP). There is a proprietary protocol known as Inter-Tel Protocol mode (ITP) as well as the industry-standards protocol SIP, MGCP, OAI, and RTP.

- Voice to/from the MiVoice Office 250
- Peer-to-peer voice
- ITP: Inter-Tel Protocol
- TFTP: Trivial File Transfer Protocol
- MGCP: Media Gateway Control Protocol
- SIP: Session Initiation Protocol

Private Networking

The MiVoice Office 250 is capable of connecting with other MiVoice Office 250 systems to form a network that supports many call processing features. This networking capability can be implemented using point-to-point T1 or PRI circuits, or it can be implemented using IP. This appendix is focused on IP-based networking. See “Private Networking“ on **page 389** for information on how to set up and program IP Networking. Appendix A also provides information on setting up and programming T1 networking.

Computer Telephony Integration (CTI)

This section discusses computer telephony integration using the Open Architecture Interface (OAI).

Audio and Web Conferencing

Audio and Web Conferencing runs on a standalone server to provide multiparty audio conferences. It utilizes ITP, OAI, and RTP, to communicate with the MiVoice Office 250 as multiple Proprietary Protocol IP phones.

System Manager

System Manager is a server-based application that centralizes management functions for the MiVoice Office 250 along with various peripheral components, such as Unified Communicator®. For complete functionality, System Manager requires significant connectivity to and from the managed devices. These requirements are outside the scope of this appendix. They are described in the *System Manager Installation and Maintenance Manual*, part number 835.2743.

Topology Requirements

With regard to network topology discussed in this appendix, keep the following requirements in mind:

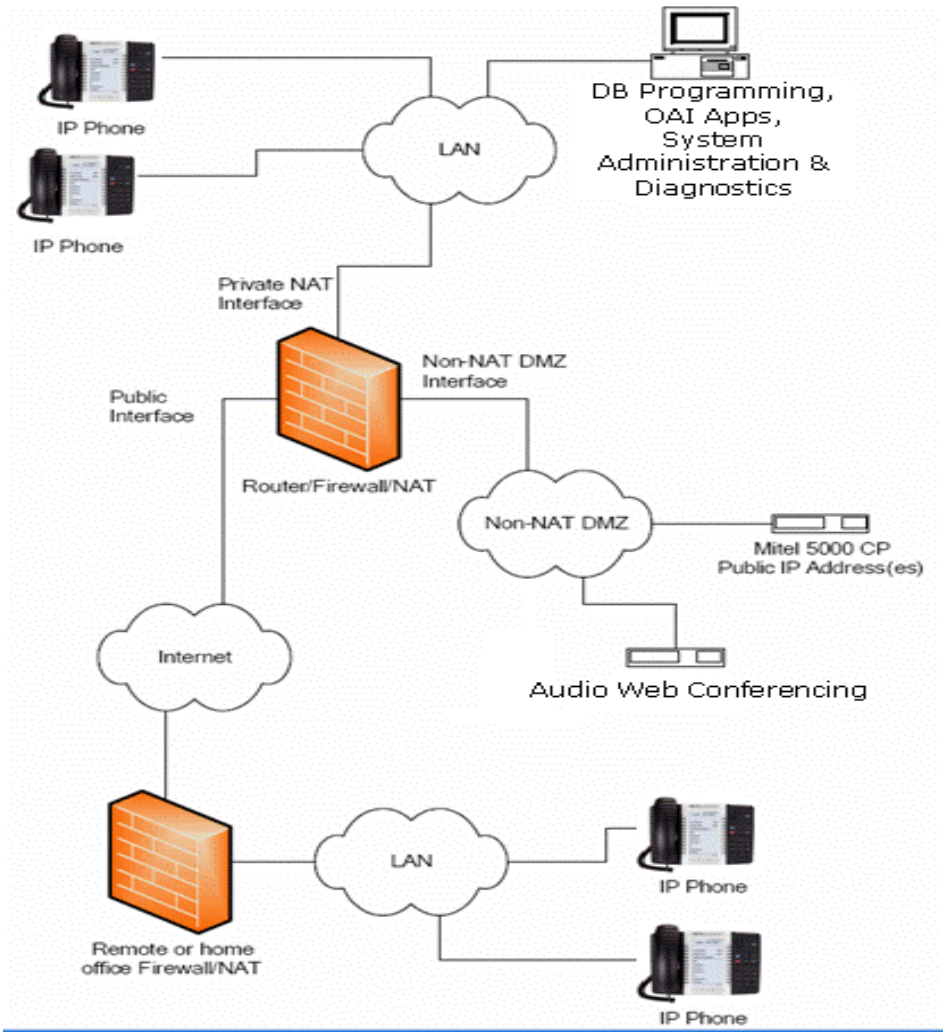
- There is only one network interface for each MiVoice Office 250.
- Private Networking cannot operate through a NAT device.
- The communication between SIP phones and the MiVoice Office 250 must not pass through near-end NAT, unless it is performed by a SIP-aware firewall. The same requirement exists for a firewall performing far-end NAT.
- The communication between the MiVoice Office 250 and AWC server must ***not*** pass through NAT.

Private (Trusted) Networks

With regard to the last bullet in “Topology Requirements“ on **page 424**, all configurations require that the MiVoice Office 250 and AWC are on the same network. In the simplest form, all IP devices, including phones, are on the same network as shown in the following illustration.

IMPORTANT	Although VPN links avoid the security and NAT addressing issues, the encryption of VPN traffic sometimes introduces significant delays, making it unsuitable for VoIP communications. This is a function of the network connections and the networking equipment.

A Wide Area Network (WAN) provides the ability to connect two or more networks without necessarily passing through a firewall/NAT. It essentially makes one larger private network. A Virtual Private Network (VPN) uses public network facilities to provide private WAN connections. In either implementation, there are no restrictions on topology as long as all IP devices are on the same private network as shown the following illustration.



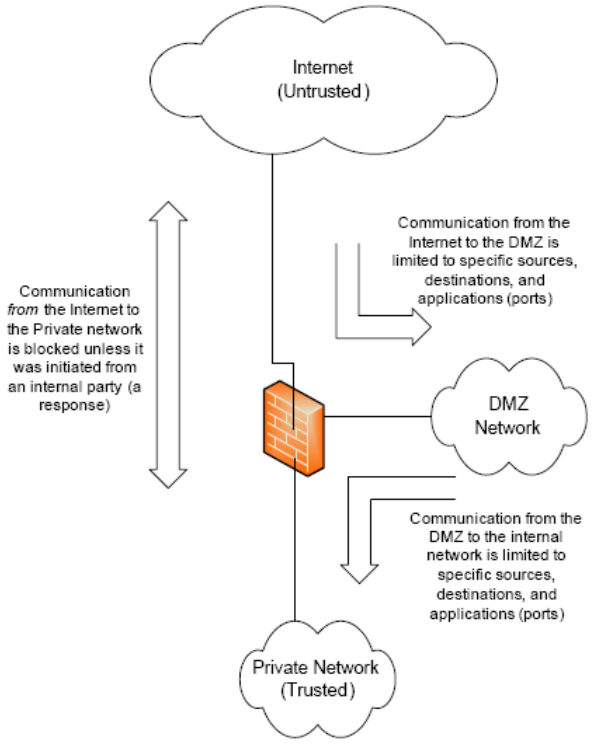
To allow IP phones on the Internet in the private networking scenario described above, you may want to consider the following options:

- Configure a static NAT assignment in the near-end NAT. Program the MiVoice Office 250 to know its NATed address along with which phones are connecting from the Internet. See “Configuring Near-End NAT Traversal for IP Phones” on **page 1-431**.
- Provide VPN connectivity between the private network and the network containing the IP phone.
- Network an additional MiVoice Office 250 using PRI networking, and place it in a non-NATed DMZ as described in “Public (Untrusted) Networks” on **page 1-427**.

Public (Untrusted) Networks

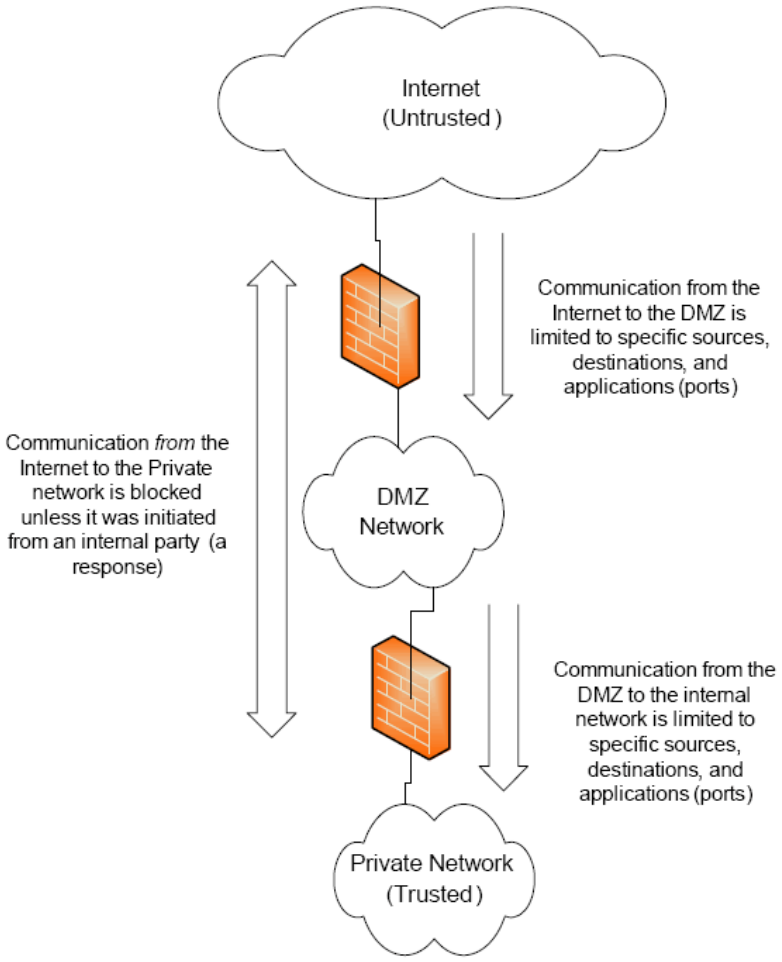
The private networking scenarios discussed in “Private (Trusted) Networks” on **page 1-425** impose very few restrictions on the topology of the IP telephony equipment because there are no security barriers between devices. However, one of the major benefits of IP telephony is the ability to leverage the existence of large public IP networks (for example, the Internet) to extend the communication network. Using public (untrusted) networks requires security measures to protect the private (trusted) network. This protection is usually provided through a firewall. Unfortunately, the functionality provided by a firewall impedes the communications necessary for IP telephony.

To use the public network in safe manner, it is necessary to provide some protection while allowing the necessary communications. In network terminology, this is usually configured using a topology known as a DMZ (DeMilitarized Zone)[1]. Devices in a DMZ network are accessible from the Internet in very specific ways according to the source and destination addresses and the applications that they support. However, the devices in the DMZ have very limited access into the private LAN to protect it should these devices be compromised. Devices in a DMZ network generally include web servers, mail servers, etc.



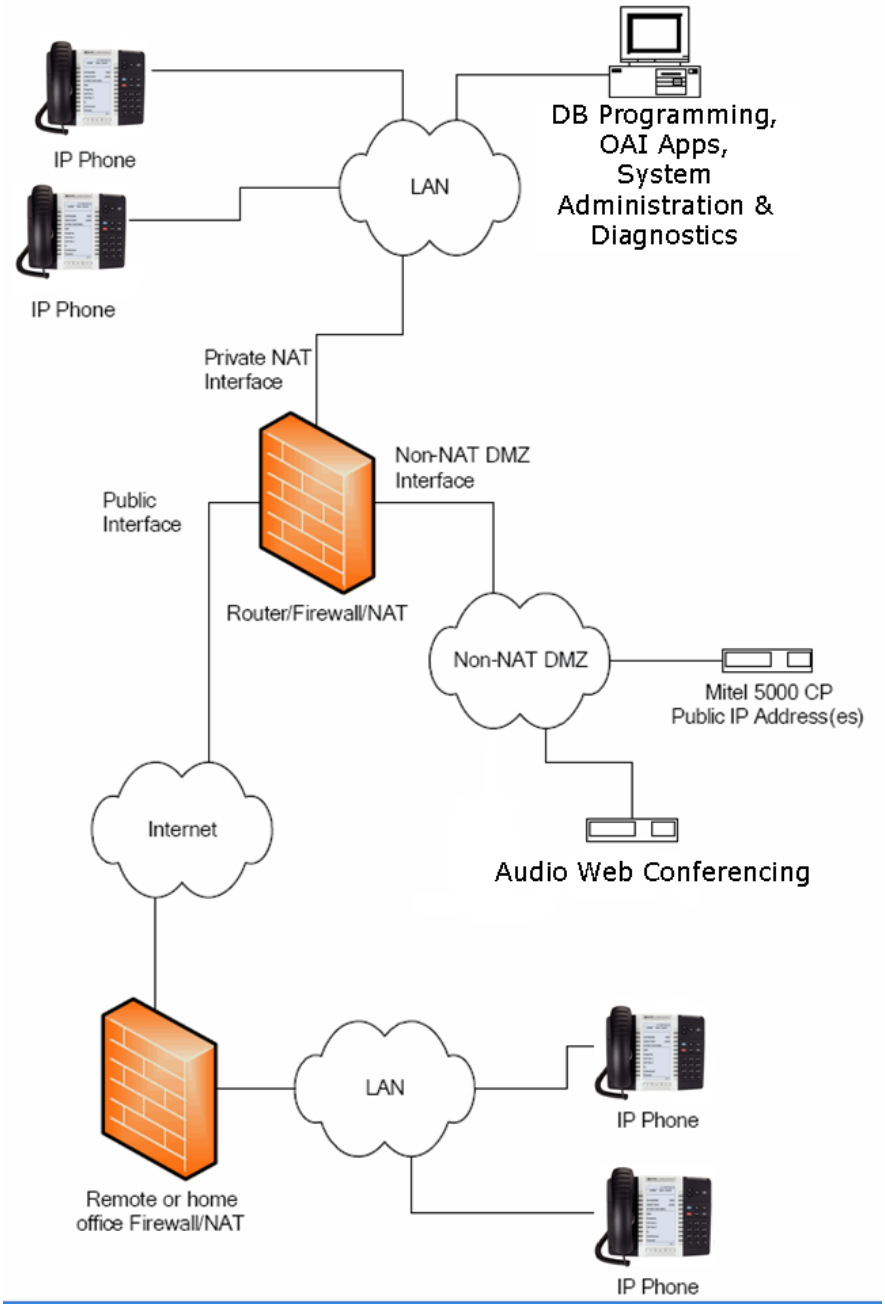
In some environments, DMZs are implemented in what is referred to as a two-tier architecture using two firewalls as shown in the following example. In this environment, the DMZ is between the inside trusted network and the outside untrusted network. The security issues are still the same, but the configuration is split between the two firewalls.

1. Many home router/firewalls (for example Linksys®) inaccurately use the term DMZ to describe something else: a internal host accessible at a public address. This appendix uses the term DMZ as described in “Terminology” on **page 1-418**.

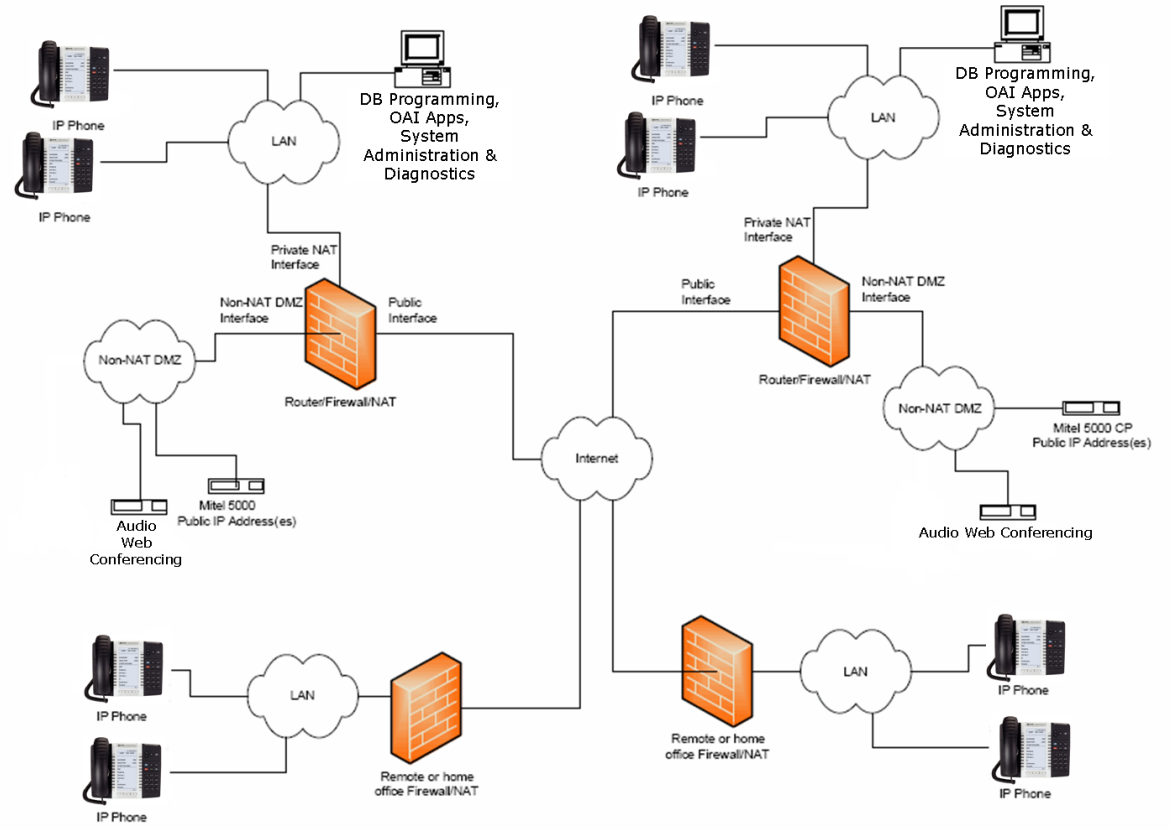


In addition to providing protection, firewalls also perform a Network Address Translation (NAT) function (described in “Terminology” on **page 1-418**) in which private IP addresses are converted to public IP addresses as packets go out through the firewall. As a general rule, IP telephony devices do not perform well when the communications must pass through a device performing NAT.

For the MiVoice Office 250 and other IP devices to function properly in the DMZ, *the DMZ must be a non-NAT network*. That is, the devices in the DMZ use *public* IP addresses. The firewall can still provide some protection from the public Internet, but it does not translate the IP addresses or port numbers. An example is shown in the following illustration.

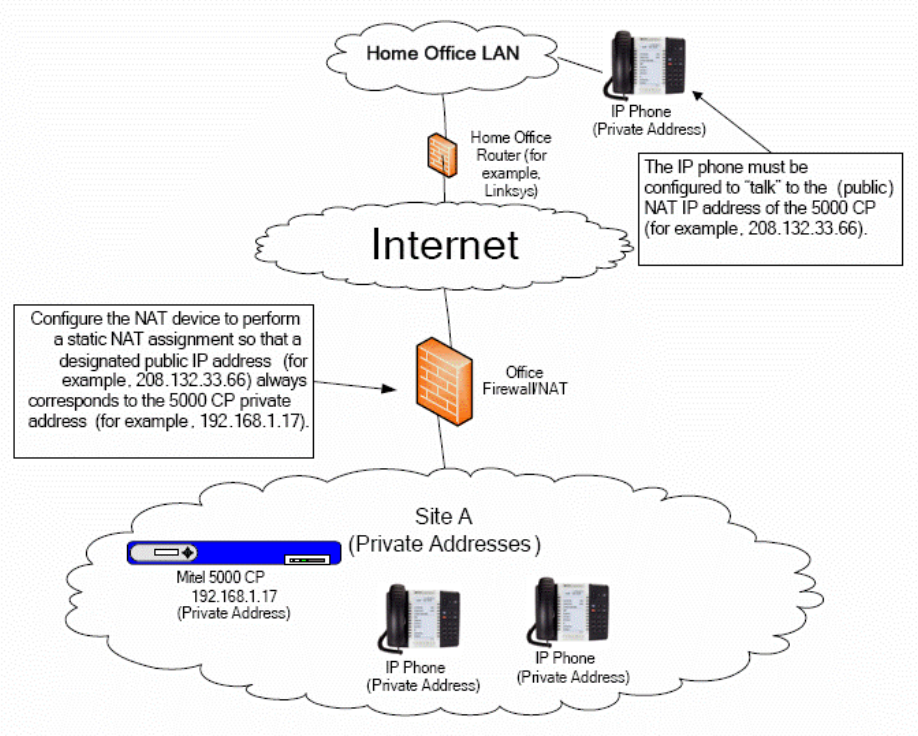


This topology uses public addressing. To support IP addressing means that networking must be routed over the Internet. To use Private Networking, set up each location in the same manner. An example is shown in the following illustration.



Configuring Near-End NAT Traversal for IP Phones

The MiVoice Office 250 supports traversing near-end NAT using statically assigned NAT addresses. An administrator configures the NAT device to always associate a specific public address with the MiVoice Office 250 native address. This is also known as “Full Cone NAT.” A system equipped with an optional PEC-1 has two IP addresses, so it requires two static NAT assignments. The static NAT assignment and the IP phone configuration are illustrated below.



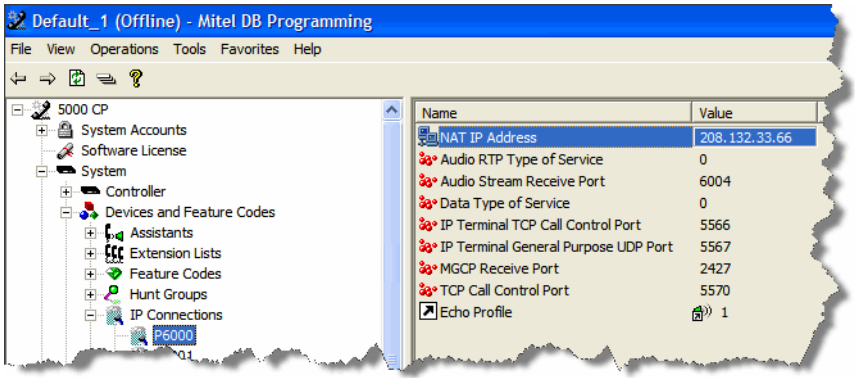
Base Server

To set up Near-End NAT Traversal for IP Phones on a Base Server-only system:

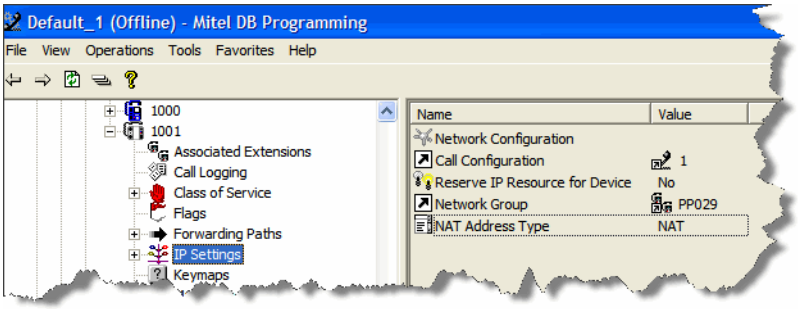
1. Configure the firewall/NAT device for a static mapping. This associates a public IP address with the MiVoice Office 250 native private address.

IMPORTANT	Configure the firewall security settings for the newly mapped IP address(es) as described in “Port Usage and Protocols” on page 442. Depending on how the particular firewall defaults its security, failure to configure the security may result in no-audio conditions or security vulnerabilities.

2. In the DB Programming IP Connections folder, program the NAT IP Address to be the public address that was associated with the native private address in **step 1**.

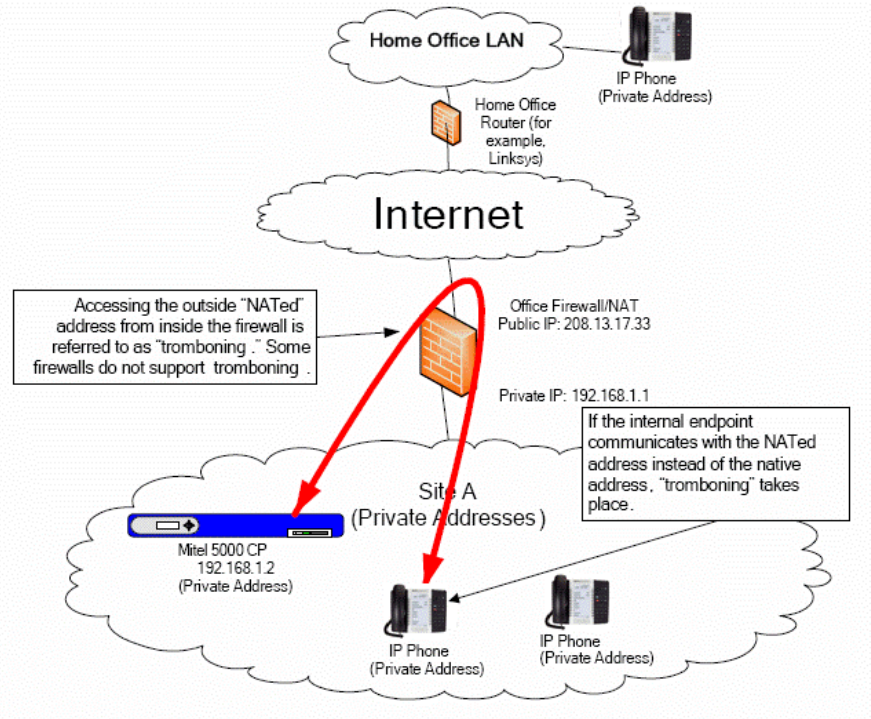


3. In DB Programming, create an IP phone and specify the MAC address.
4. In the IP Settings folder for the newly created IP phone from the Internet, set the NAT Address Type to **NAT**, as shown in the example below.



5. Configure the IP phone (that is on the Internet) to communicate with the Base Server (public) NAT address. You can do this through the IP phone setup sequence or through the IP phone Web page. The address to use is the public address associated with the Baser Server main IP address – regardless of whether the system is equipped with an optional PEC-1.

6. For the “internal” IP phones (those in the internal network with private addresses), set the NAT address type **Native**, and configure the IP phones to communicate with the Base Server (private) internal address, 192.168.1.2. Incorrect configuration may result in an undesirable effect called tromboning (also known as “hairpinning”). Not all firewalls support tromboning, so avoid this configuration, if possible. The following diagram illustrates tromboning.



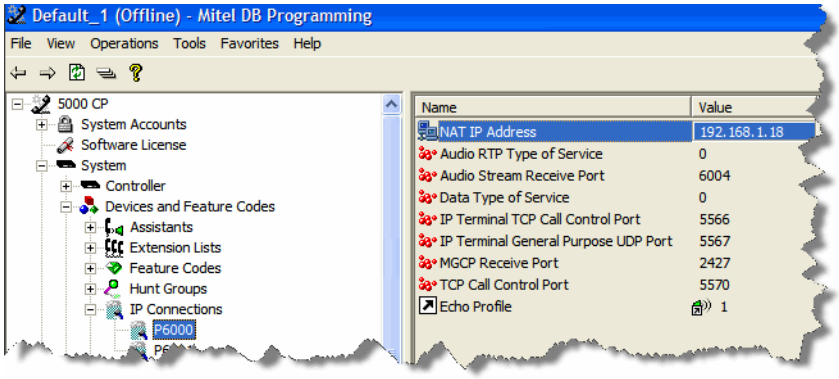
Base Server with a PEC-1

To set up Near-End NAT Traversal for IP Phones on a Base Server with a PEC-1:

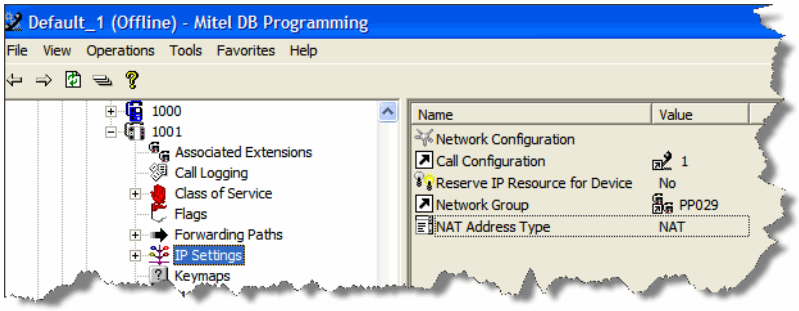
1. Configure the firewall/NAT device for a static mapping for the IP address of the Processor Expansion Card. This associates a public IP address with the native address of the Processor Expansion Card (PEC-1).

IMPORTANT	Configure the firewall security settings for the newly mapped IP address(es) as described in “Port Usage and Protocols” on page 442. Depending on how the particular firewall defaults its security, failure to configure the security may result in no-audio conditions or security vulnerabilities.

2. In the DB Programming IP Connections folder, program the NAT IP Address to be the public address that was associated with the native private address in **step 1**.



3. In DB Programming, create an IP phone and specify the MAC address.
4. In the IP Settings folder for the newly created IP phone from the Internet, set the NAT Address Type to **NAT**, as shown in the example below.



5. Configure the IP phone (that is on the Internet) to communicate with the system's (public) NAT address. You can do this through the IP phone setup sequence or through the IP phone Web page. The address to use is the public address associated with the Base Server's main IP address – regardless of whether the system is equipped with an optional PEC-1.
6. For the "internal" IP phones (those in the internal network with private addresses), set the NAT address type **Native**, and configure the IP phones to communicate with the system's (private) internal address, 192.168.1.2. Incorrect configuration may result in an undesirable effect called tromboning. Not all firewalls support tromboning, so avoid this configuration, if possible. The diagram on the previous page illustrates tromboning.

Configuring Near-End NAT Traversal for IP Gateway Trunks

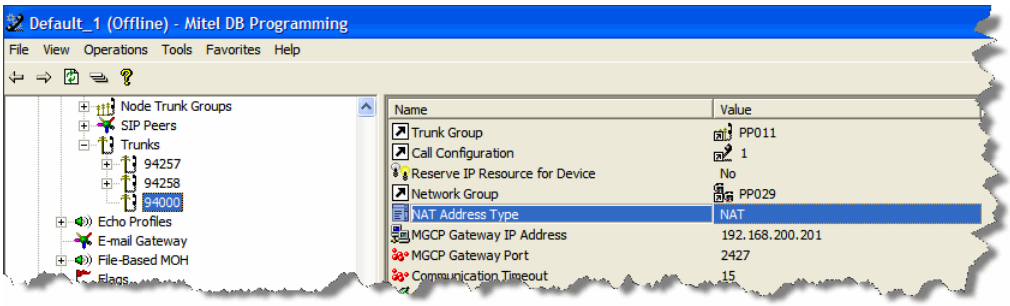
In addition to IP phones, the MiVoice Office 250 also supports near-end NAT traversal for IP trunk gateways. Note that IP trunk gateways impose additional restrictions on *far-end* NAT.

NOTICE

Systems equipped with a PEC-1 have two IP addresses; the routing information for both addresses must be configured identically. Failure to configure the IP addresses in the same subnet with the same accessibility can result in no-audio problems.

To configure near-end NAT traversal for IP Gateway trunks:

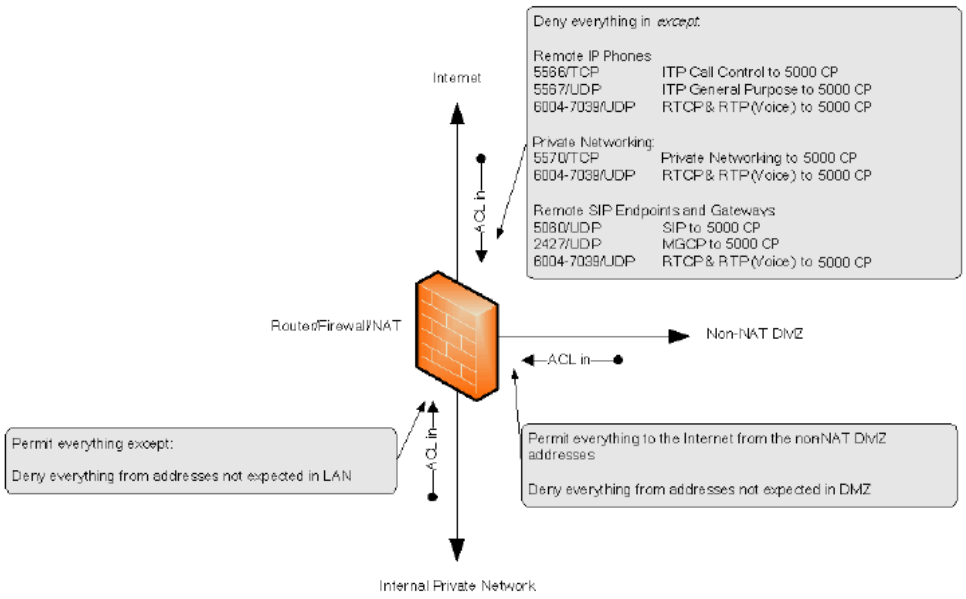
1. Configure the static NAT assignments and the MiVoice Office 250 NAT IP Address assignments as described in step 1 on **page 431** (Base Server only) or step 1 on **page 433** (Base Server with a PEC-1). This includes the information in the IMPORTANT notes.
2. For both MGCP and SIP Trunk gateways, set the NAT Address Type to **NAT**.



3. Repeat this process for each IP gateway trunk.

Firewall Configuration

In computer networks, a totally secure network is practically impossible. In general, the more types of communications allowed from the Internet, the less secure the network. Therefore, a network should be configured so as to allow only the minimum level of communication unless other capabilities are required. An example is shown in the following illustration.



In some environments, it will be necessary to allow additional types of communications through the firewall. As stated above, these communications should be allowed only when necessary. Although strong passwords are always a good idea, this is especially important when applications are opened to the Internet.

Private Networking

This is the protocol used to allow MiVoice Office 250 systems to communicate with each other. To allow Private Networking to the Internet, allow TCP port 5570.

Remote Administration

These are the protocols that provide the capability to perform certain administration activities from outside the firewall. To enable remote administration including DB Programming over the Internet, enable TCP port 44000. To further tighten security, this communication can be filtered by source address[1].

1. Source address filtering accepts communication from specific IP addresses. While source address filtering provides some additional level of security, IP source addresses are easily spoofed. In addition, source addresses are often DHCP-assigned making it impractical to know the source addresses in advance.

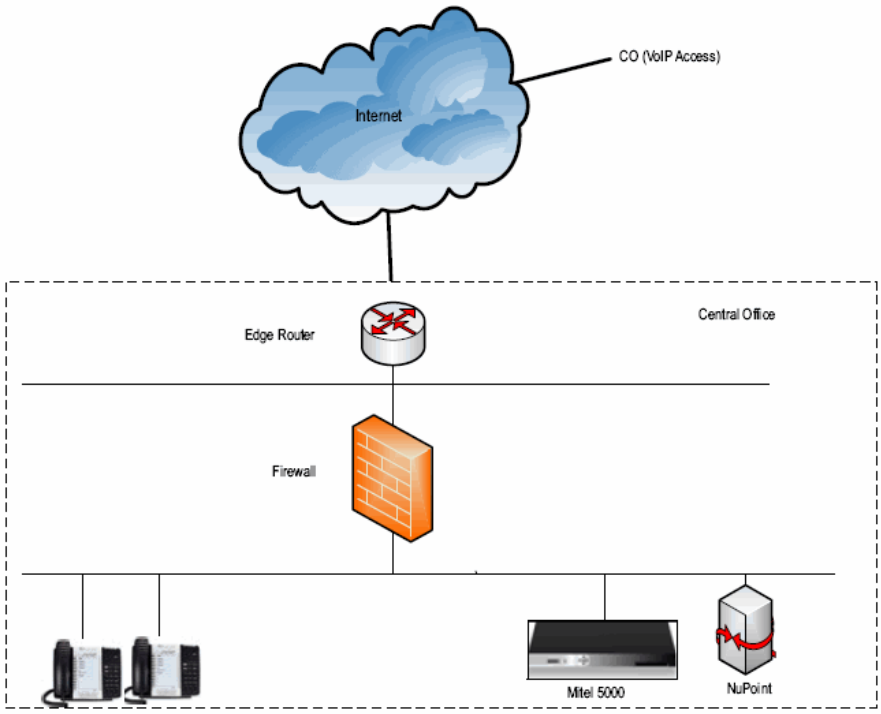
Remote Maintenance and Diagnostics

These protocols provide capabilities to diagnose problems, update software, etc. To enable remote maintenance and diagnostics over the Internet, enable the ports for DB Programming (TCP port 44000), Web interfaces (TCP port 80), SSH (TCP port 22), and AVDAP Monitor (TCP port 4444).

Near-End NAT/Firewall Traversal

Figure 12 shows a simple customer network with a NAT/firewall between the private network and the Internet. All SIP and RTP between the SIP peer (in the public network) and the MiVoice Office 250 network goes through this firewall.

Figure 12: Near-End NAT/Firewall Traversal Network Diagram



The firewall must be SIP-aware for the configuration in Figure 12 on **page 437** to work and to use service provider SIP trunks.

The following are various deployment configurations for the MiVoice Office 250 to communicate with a SIP trunk service provider in the public network. You can install the MiVoice Office 250:

- Behind a SIP-aware NAT/firewall. The SIP-aware NAT/firewall provides SIP fix-up for SIP traffic between the MiVoice Office 250 and the SIP trunk service provider network. This configuration requires the MiVoice Office 250 to register with the SIP peer.
- Behind a corporate enterprise NAT/firewall provided, you create a static mapping on the NAT/firewall. This static mapping must map the public IP address of the MiVoice Office 250 to the private IP address of the MiVoice Office 250. Also, there must not be any PAT on the NAT/firewall. This configuration does not require the MiVoice Office 250 to register with SIP peer.

- With a public IP address, so that there is no NAT/firewall between the MiVoice Office 250 and the SIP peer. This configuration does not require the MiVoice Office 250 to register with the SIP peer.

NAT Settings

The following scenarios describe how to configure the network for the MiVoice Office 250 to communicate with the SIP peer. You must program the NAT settings in Database (DB) Programming for the SIP peer trunk group based on the firewall. Refer to the *MiVoice Office 250 Features and Programming Guide*, part number 580.8006.

Scenario 1 – Recommended (SIP-Aware Firewall)

The MiVoice Office 250 is inside the corporate network and there is a SIP-aware firewall between the MiVoice Office 250 and the SIP peer. This is the recommended configuration. Set the NAT Type for the SIP peer to “No NAT or SIP-Aware NAT.”

Scenario 2 – Recommended (not SIP-Aware Firewall)

The MiVoice Office 250 is inside the corporate network and the firewall between the MiVoice Office 250 and the SIP peer is not SIP-aware. Set the NAT Type for the SIP peer to “Non-SIP-Aware NAT.” The other settings for this configuration are as follows:

Base Server Only

The firewall should have a static mapping for the Base Server: Set the:

- “System NAT IP Address” in System\IP Settings in DB Programming to the statically mapped public IP address of the Base Server.
- “NAT IP Address” in System\Devices and Feature Codes\IP Connections\<*node*>\P6000 to the statically mapped public IP address of the Base Server.

Base Server with a PEC-1

The firewall should have two statically mapped public IP addresses for the MiVoice Office 250 – one each for the Processor Module and the Expansion Card. Set the:

- “System NAT IP Address” in System\IP Settings in DB Programming to the statically mapped public IP address of the Processor Module.
- “NAT IP Address” in System\Devices and Feature Codes\IP Connections\<*node*>\P6000 to the statically mapped public IP address of the Processor Module.
- “NAT IP Address” in System\Devices and Feature Codes\IP Connections\<*node*>\P6001 to the statically mapped public IP address of the Expansion Card.

Base Server with a PEC-1 and a PS-1

The firewall should have three statically mapped public IP addresses for the MiVoice Office 250– one each for the Processing Server, the Processor Module, and the Processor Expansion Card. Set the:

- “System NAT IP Address” in System\IP Settings in DB Programming to the statically mapped public IP address of the Processing Server.
- “NAT IP Address” in System\Devices and Feature Codes\IP Connections\<*node*>\P6000 to the statically mapped public IP address of the Processor Module.
- “NAT IP Address” in System\Devices and Feature Codes\IP Connections\<*node*>\P6001 to the statically mapped public IP address of the Expansion Card.

Scenario 3 – Not Recommended

The MiVoice Office 250 is in the public network with no NAT between the MiVoice Office 250 and the SIP peer. This is not a recommended configuration for security reasons. However, if it is done, then set the NAT Type for the SIP peer to “No NAT or SIP-Aware NAT.”

Near-End NAT/Firewall Configuration Recommendations

The following are some near-end NAT/firewall recommendations:

- Create a static mapping for the MiVoice Office 250.
- If you use an unsupported SIP-aware firewall or NAT device, you must disable peer-to-peer traffic between any Mitel component (except the MiVoice Office 250) and the SIP trunk. All traffic to or from the SIP trunk terminates at the MiVoice Office 250.

Firewall Configuration in MiVoice Office 250 Admin Web Portal

Security Updates

New OpenSSL Version

There is a new OpenSSL version 1.0.2l.

Firewall Improvements

Improvements have been made to the existing **Firewall**, including the ability to **Save** and **Restore** settings. Firewall Configuration options have been extended to include the PS-1.

This also includes fixes for system reset/lockup issues that happen due to specific network activity. These systems use remote IP and MiNet Phones remotely and were experiencing excessive TCP Duplicate ACK Message storms.

Extensive port scanning applications can also lead to a high packet rate that can slow down and freeze the system.

The programming of **Firewall Rules** is only available in the A**dministrator Web Portal**. System defined rules are grouped by their purpose, into:

- initial_filtering
- tcp_dup_acks
- port_scanning chains

These rules cannot be modified. The User can only Enable or Disable the full set of rules.

Supported Firewalls

Table 73 lists the SIP-aware firewalls that have been tested and are supported by Mitel. For recommended configurations, refer to the Mitel Knowledge Base Web site (accessible through Mitel Online) for details.

Table 73: List of Tested and Supported SIP-Aware Firewalls

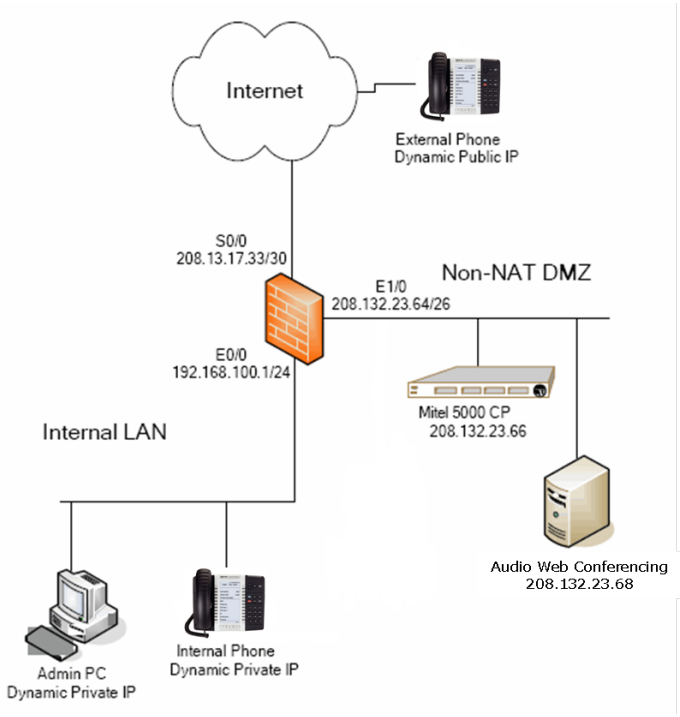
Vendor Name	Software Version	SIP-Aware
Cisco® ASA	V7.0 or later	Yes
Juniper® Networks Netscreen-25[a]	5.4.0u1.0	Yes
SonicWALL® Pro 2040[a]	SonicOS Standard 3.1.0.7-77s	Yes
ADTRAN Total Access 908e[b]	A1.03.00.E	No

a. This firewall does not translate the private IP address in the “P-Asserted Identity” field of the SIP header.

b. This firewall acts as a SIP gateway and interfaces with the SIP peer.

Internal Session Initiation Protocol (SIP)

The SIP protocol presents some special challenges in that it does not function well when passing through NAT. The only solution to this problem is to utilize a SIP-aware firewall that can perform the appropriate adjustments associated with NAT.



Port Usage and Protocols

Stateful Packet Inspection (SPI) is required when a MiVoice Office 250 is placed in a non-NATed DMZ in a Private Networking scenario and is communicating over the Internet.

A port is a form of addressing that is subordinate to IP addressing. As such, port numbers serve to distinguish one protocol or application from another at the same IP address. For example, the primary protocol that browsers such as Internet Explorer use to access Web pages is HTTP, and HTTP uses port TCP 80. A single device with a single IP address can support many protocols simultaneously by using different port numbers.

Separate sets of ports exist for Transfer Control Protocol (TCP) and User Datagram Protocol (UDP). For port-assignment details, see “UDP Port Assignments” on **page 1-444**.

Firewalls are devices used to promote network integrity and security by filtering data packets based on IP addresses and port numbers. For example, suppose a firewall is configured to allow communication with a Web server, which uses HTTP and TCP port 80. This configuration would prevent the server from being accessed as a mail server, which uses SMTP and TCP port 25.

For security reasons, all ports should be disabled except those that are specifically required. The default state for many routers is to disallow everything, but this practice varies from one manufacturer to another.

NOTE This appendix assumes that everything is disallowed unless specifically allowed.

A firewall is designed to prevent access from an untrusted network such as the Internet while allowing communication initiated from within a trusted network. This capability is referred to as Stateful Packet Inspection (SPI) when the firewall is configured to keep track of the state of each session.

When enabled, SPI makes the firewall remember that an outbound TCP connection was initiated from behind the firewall and allows packets associated with this connection back through the firewall.

IMPORTANT	If: You use Private Networking over the Internet. and You have placed the MiVoice Office 250 in a non-NATed DMZ to provide port-filtering protection from Internet threats. Then: In addition to opening the ports specified in **Table 74**, you must also enable SPI between the Internet and the DMZ.

The port numbers listed on the following page are system defaults for the MiVoice Office 250, but many of them are configurable by the system administrator.

Except where otherwise noted, these are ports on which the MiVoice Office 250 can listen for new connections. The MiVoice Office 250 may also establish outbound connections with random source ports. For proper operation, the network should not limit outbound connections from the MiVoice Office 250 or responses to these established outbound connections.

Table 74: Port Assignments

Type	Default Ports and Protocol	Comments for Firewall
Voice Over IP	6004–7038/UDP(RTP) 6005–7039/UDP(RTCP)	Enable only if using VoIP outside the firewall.
Inter-Tel Protocol (ITP)	5566/TCP(ITP Call Control) 5567/UDP(ITP Call Control)	Enable only if using ITP IP phones outside the firewall. Phones that use ITP include the 86xx IP phones and IP-SLA.
SIP	5060/UDP	Enable only if using SIP phones, trunks, or gateways outside the firewall.
MGCP	2427/UDP	Enable only if using MGCP trunks outside the firewall.
Private Networking[a]	5570/TCP	The MiVoice Office 250 uses this port for inter-node communication. This protocol does not support NAT on either side. Private Networking over IP will not work through a firewall with NAT enabled.
Administration/Applications — Unsecured (prior to v4.0)		
• DB Programming • OAI • Diagnostic Monitor • System Manager	4000/TCP	The MiVoice Office 250 uses these ports for system administration, diagnostics, applications, and upgrades. Enable only if necessary for external communications through the firewall.
• OAI Broadcast	4000/UDP	
• Web Interface	80/TCP(HTTP)	
Administration/Applications — Secured (v4.0 and later)		
• System Administration & Diagnostics • OAI • DB Programming • Message Print • System Monitor	44000/TCP	The MiVoice Office 250 uses these ports for system administration, diagnostics, applications, and upgrades. Enable only if necessary for external communications through the firewall.
• Web Interface	443/TCP(HTTPS)	
Diagnostics		
• Diagnostics Web Interface	80/TCP(HTTP)	Enable only if directed by Mitel support personnel. (AVDAP Monitor is used for certain voice mail diagnostics. Secure Shell is used for system diagnostics and communication between the Base Server and the Processing Server.)
• AVDAP Monitor	4444/TCP	
• Secure Shell	22/TCP(SSH)	
Voice Mail	25/TCP(SMTP)	The MiVoice Office 250 uses SMTP to receive VPIM messages.

(Sheet 1 of 2)

Table 74: Port Assignments (Continued)

Type	Default Ports and Protocol	Comments for Firewall
Dynamic Host Configuration Protocol (DHCP)	67 and 68/User Datagram Protocol (UDP)	The MiVoice Office 250 and IP phones use DHCP to obtain their IP address, subnet mask, default gateway, TFTP server, and ICP address. It is optional if all network information is statically programmed.
Trivial File Transfer Protocol (TFTP)	20001,69/UDP	The phone uses TFTP to download firmware and applications from the MiVoice Office 250 during the boot process. The phone/source port number may be random, but the MiVoice Office 250 destination uses the specified ports. The phone ships with the port setting at 0 (recommended). When left at this setting, the phone tries port 20001 followed by port 69.
MiNet	6800–6802/Transfer Connection Protocol (TCP)	The MiNet protocol is for basic call control for Mitel IP phones.
Switch Application Communication (SAC)	3998 and 3999/TCP	The SAC protocol is for phone applications and programmable buttons on Mitel 53xx IP phones.
RTP (86xx phones)	5004–5007/UDP	Real-Time Transfer Protocol (RTP) is for streaming audio. The MiVoice Office 250 and IP phones use the specified ports to send to and receive from the peer phone. RTP (86xx phones) - FOR AUDIO TOWARDS THE PHONE 5004 - 5007/UDP RTP (52xx/53xx phones) FOR AUDIO TOWARDS THE PHONE 50098 - 50508/UDP RTP (Processor Module) FOR AUDIO TOWARDS THE 5000 6004 - 6261/UDP RTP (Processor Expansion Card - ONLY REQUIRED IF PEC INSTALLED) FOR AUDIO TOWARDS THE 5000 6604 - 7039/UDP
RTP (52xx/53xx phones)	50098–50508/UDP	
RTP (Processor Module)	6004–6261/UDP	
RTP (Processor Expansion Card)	6604–7039/UDP	

(Sheet 2 of 2)

a. Private, managed networks are preferred for Private Networking. Although networking over the Internet is possible, it is not recommended.

UDP Port Assignments

Expanded IP phone capability with v2.0 and later requires changes in User Datagram Protocol (UDP) port usage. If the MiVoice Office 250 v2.x or later equipment operates inside a firewall and services external IP phones, the firewall configuration needs to be updated to reflect the changes in the UDP port assignments of system.

Tutorial

In DB Programming, the Audio Stream Receive Port defines the first of 33 even port numbers that off-node IP resources use to transmit audio data to this IP resource. You must choose an even-numbered value. The base audio port number, plus 64 additional port numbers, cannot conflict with other port numbers on the IP connection.

For example, if the base audio port number is 5000, no other port numbers on the IP connection can fall in the range from 5000 to 5064. Changing the Audio Stream Receive Port resets the IP resources. The system broadcasts changes to an IP connection Audio Stream Receive Port to the other nodes in the private network as a database update. This field corresponds to the off-node IP connection Remote Audio Receive Port. To change the port number, enter the new number in the text box, then press **ENTER** or click on another field. The default is *5004,* and the range of values is *1026–65468*.

UDP Port Assignments in v2.x and Later

To support external IP phones, the system requires that a range of UDP ports be opened through the firewall. **Table 74** shows the UDP port range applicable to v1.x and the UDP ports applicable to v2.0 and later. These port assignments apply to the Processor Module (PM-1) and the Processor Expansion Card (PEC-1) in the Mitel 5000, but *not* to the Processing Server PS-1.

Knowledge Base Article: Port Information (KB2377)

The following is a summary of KB article 2377:

With the release of software version 1.1.X.XXX, NAT functionality was implemented into the Mitel® 5000.

We advise a one to one static NAT (all traffic to/from the public to the private address is forwarded unchanged). If you wish to open only specific ports to the private IP address then these are the necessary ports:

- 5566 TCP
- 5567 UDP
- 6004 - 6247 UDP

In the 2.0.x.xxx software release the number of total IP phones was increased, the UDP port range used by the system also has increased as below. The functionality and programming of the system as far as NAT is concerned has not changed though.

- 5566 TCP
- 5567 UDP
- 6004 - 6604 UDP for systems without a PEC-1 or PS-1
- 6004 - 7039 UDP for systems equipped with a PEC-1 or PS-1
- Processor Module (PM-1) base port – 6004
- Processor Expansion Card (PEC-1) base port – 6604

Systems equipped with a PS-1 should be treated systems equipped with a PEC-1 as far as NAT is concerned. You will need IP addresses for both the PM-1 and PEC-1. See KB 3352 for further details on connecting IP devices to systems equipped with a PS-1.

Notable Differences with the MiVoice Office 250Platform

Three notable differences exist with the MiVoice Office 250 platform:

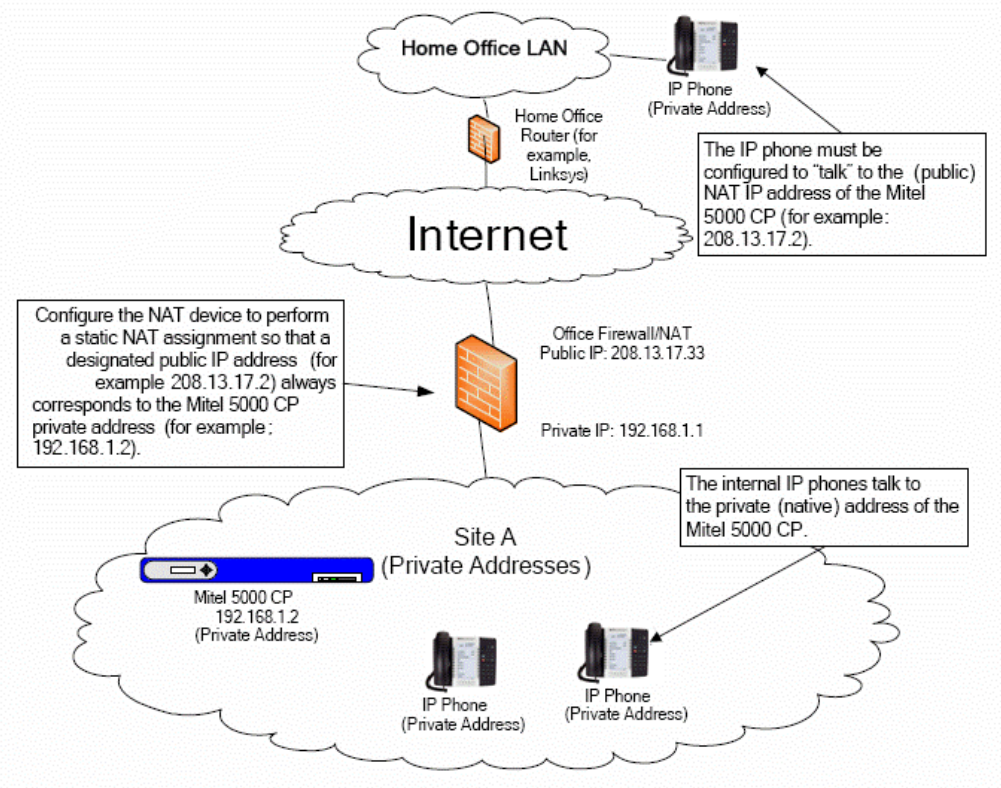
- The MiVoice Office 250 (no matter the size) has a single physical Ethernet port shared for all IP telephony and administrative functions. Also, 5000 systems equipped with a PEC-1 have two IP addresses. The second IP address is used for VoIP only.
- Because the MiVoice Office 250 has a single physical Ethernet interface, it has a single network location. It cannot operate both inside the firewall and outside the firewall.
- The MiVoice Office 250 is built on an open-source Linux® platform that could present a greater security risk than other platforms, which may operateson proprietary technology.

Example Cisco Configuration Commands

The following commands are provided as an example to further illustrate the recommended configuration with respect to security. QoS configuration is beyond the scope of this appendix. These configuration examples are based on the Cisco IOS Firewall feature set. Adtran® routers use a very similar command set called AOS. For more information about Adtran routers Adtran Netvanta Switch/Router on **page 1-452**. These commands are not intended to be complete; they are only a guideline.

Private Network With Near-end NAT Traversal

The following diagram shows a configuration example.



The following commands set up the connection to the internal LAN. NAT is enabled, and it uses **`access-group e0in`** for traffic coming in (to the router) from the internal LAN.

```
interface Ethernet0/0
 description Internal LAN
 ip address 192.168.1.1 255.255.255.0
 ip access-group e0in in
 ip nat inside
 ip inspect inspector in
!
```

- The following section defines the access control list (the rules) for traffic coming from the internal LAN into the router. As a general rule here, you want to allow about everything to go out from a trusted LAN.

```
! Access Control List e0in
!
ip access-list extended e0in
 permit ip 192.168.1.0 0.0.0.255 any
 deny   ip any any
```

- The following section sets up the connection to the Internet. NAT is enabled between the Internet and the internal LAN. Traffic from the Internet is filtered using the access-group called *Internet*.

```
interface Serial0/0
 description connected to Internet
 ip address 208.13.17.33 255.255.255.252
 ip access-group s0in in
 ip nat outside
```

- The following section defines the access control list (the rules) for traffic coming from the Internet to either the Internal LAN or the DMZ. This is the first line of defense, so you want to filter as much as possible. Responses to communications initiated from inside (for example, http request for a web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.13.17.2 eq 5566
 permit udp any host 208.13.17.2 eq 5567
 permit udp any host 208.13.17.2 range 6004 6247
 deny   ip any any
```

The following command associates the public address with the MiVoice Office 250 private address. This creates the “NATed” address. The ISP must be known to send packets for this public IP to the MiVoice Office 250.

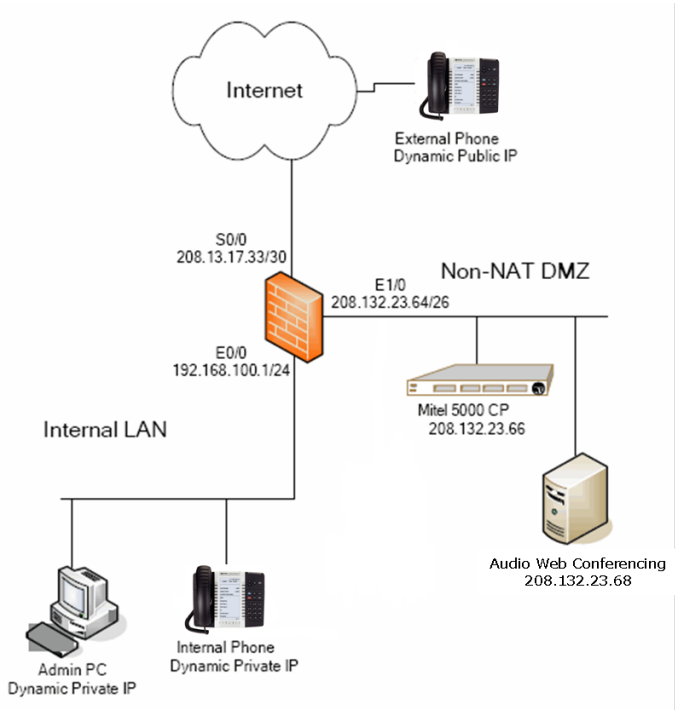
```
ip nat inside source static 192.168.1.2 208.13.17.2
!
```

The following commands set up the dynamic NAT configuration to use a pool of public addresses. This is not specific to this example, but it is included here to contrast static vs. dynamic NAT.

```
ip nat translation timeout 86400
ip nat translation tcp-timeout 86400
ip nat translation udp-timeout 300
ip nat translation dns-timeout 60
ip nat translation finrst-timeout 60
ip nat pool NatPool0 208.13.17.40 208.13.17.62 netmask 255.255.255.0
ip nat inside source list 1 pool NatPool0 overload
!
```

Non-NAT DMZ Configuration

The following illustrates a nonNAT DMZ configuration.



Single Node with ITP Phones

In this example you use the following commands set up the connection to the internal LAN. In these commands, NAT is enabled and it uses `access-group e0in` for traffic coming in (to the router) from the internal LAN.

```
interface Ethernet0/0
 description Internal LAN
 ip address 192.168.100.1 255.255.255.0
 ip access-group e0in in
 ip nat inside
 ip inspect inspector in
!
```

- The following section defines the access control list (the rules) for traffic coming from the internal LAN into the router. As a general rule, this example allows just about anything to go out from the trusted LAN.

```
! Access Control List e0in
!
ip access-list extended e0in
 permit ip 192.168.100.0 0.0.0.255 any
 deny   ip any any
```

- The following section sets up the connection to the Internet. NAT is enabled between the Internet and the internal LAN. Traffic from the Internet is filtered using the access-group called **`Internet`**.

```
interface Serial0/0
 description connected to Internet
 ip address 208.13.17.33 255.255.255.252
```

```
 ip access-group s0in in
 ip nat outside
```

- The following section defines the access control list (the rules) for traffic coming from the Internet to either the Internal LAN or the DMZ. This is the first line of defense, so filter as much as possible. Responses to communications initiated from inside (for example, http request for a Web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit udp any host 208.132.23.66 range 6004 6247
 deny   ip any any
```

- The following section sets up the connection to the DMZ. NAT is not enabled between the Internet and the DMZ. Traffic from the Internet is filtered using the access-group called DMZ. The “inspect” statement enables the stateful firewall functionality.

```
interface Ethernet 1/0
 description Site DMZ LAN
 ip address 208.132.23.66 255.255.255.192
 ip inspect dmzinspector in
 ip access-group e1in in

 ip inspect name dmzinspector udp
 ip inspect name dmzinspector tcp
 ip inspect name dmzinspector sip
```

- The following section defines the access control list (the rules) for traffic coming from the DMZ to either the Internal LAN or the Internet. Limit the communications between the DMZ and the internal LAN as much as possible in the event one of the DMZ nodes is compromised.

```
ip access-list extended e1in
 deny   ip any 192.168.100.0 0.0.0.255
 permit ip any any
!
```

ITP Phones and Networking

This example shows that to add support for networking, you expand the ACL to allow the Private Networking port to be accessible from the Internet to the MiVoice Office 250. Responses to communications initiated from inside (for example, http request for a Web page) are controlled by the firewall functionality through dynamic ACLs.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit udp any host 208.132.23.66 range 6004 6247
 deny   ip any any
```

ITP Phones, Networking, and Remote Admin/Diagnostics

This example shows that to allow remote administration and diagnostics, you open up additional ports shown in bold in the following example. Each of these ports has a distinct function, so they should be evaluated individually to determine the need.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit tcp any host 208.132.23.66 eq 4000
 permit tcp any host 208.132.23.66 eq 4444
 permit tcp any host 208.132.23.66 eq 80
 permit tcp any host 208.132.23.66 eq 22
 permit udp any host 208.132.23.66 range 6004 6247
 deny   ip any any
```

ITP Phones, Networking, Remote Admin/Diagnostics, and SIP Phones

This example shows that to allow SIP phones from the Internet, you open up the SIP port on the 5000 or the PS-1. 53xx phones use different ports (6800, 3999). TFTP port (UDP 69; 20001 is the proprietary/primary port). SIP phones on the “internal LAN” may require additional “fixup” if NAT is performed between the internal LAN and the DMZ.

```
ip access-list extended s0in
 permit tcp any host 208.132.23.66 eq 5566
 permit udp any host 208.132.23.66 eq 5567
 permit tcp any host 208.132.23.66 eq 5570
 permit tcp any host 208.132.23.66 eq 4000
 permit tcp any host 208.132.23.66 eq 4444
 permit tcp any host 208.132.23.66 eq 80
 permit tcp any host 208.132.23.66 eq 22
 permit udp any host 208.132.23.67 eq 5060
 permit udp any host 208.132.23.66 range 6004 6247
 deny   ip any any
```

Adtran NetVanta® Switch/Router Example

Adtran® provides a family of products (NetVanta®) that are a combination of switch, router, and firewall[1]. The security considerations are independent of the router vendor, but the implementation of the DMZ using Adtran NetVanta products is slightly different. Instead of simply designating an Ethernet port on a router as the DMZ, you must first create a new VLAN containing one or more Ethernet ports. The VLAN creates a logical partition of the switch that can only communicate with the rest of the switch using the router/firewall functionality. Each set of VLAN Ethernet ports function as a single router interface. In addition, the following example makes use of access policies that allow you to apply multiple access control lists to a specific interface.

ITP Phones and Private Networking

The following example illustrates the configuration for ITP phones and Private Networking. The following commands define the VLANs within the switch. This is necessary to be able to route/firewall between the LAN and the DMZ.

```
vlan 1
  name "Default"
vlan 2
  name "DMZ"
!
```

- The following commands set up the Ethernet switch ports to belong to specific VLANs. In this example, take the first two ports and associate them with the DMZ VLAN.

  ```
  interface eth 0/1
    no shutdown
    switchport access vlan 2
  !
  interface eth 0/2
    no shutdown
    switchport access vlan 2
  !
  ```

  Set up some of the ports in the default VLAN. This example shows two ports.

  ```
  interface eth 0/3
    no shutdown
    switchport access vlan 1
  !
  interface eth 0/4
    no shutdown
    switchport access vlan 1
  !
  ```

- Each VLAN functions as a port on the router with its own IP address. Also, apply access policies to each interface. The access policies are defined further below.

  ```
  interface vlan 1
    ip address  192.168.1.1  255.255.255.0
  ```

1. Although the Adtran AOS is common among all of their switch/router platforms, this example is based on an Adtran NetVanta 1224R.

```
  access-policy Private
  no shutdown
interface vlan 2
  ip address  208.132.23.64  255.255.255.192
  no shutdown
!
```

In this example, there is a connection to the Internet using a T1 interface. Set up the IP address and apply the access policy (which is defined further below).

```
interface t1 1/1
  clock source line
  tdm-group 1 timeslots 1-24 speed 64
  ip address  208.13.17.33  255.255.255.252
  access-policy Public
  no shutdown
```

- The following commands define access lists for different kinds of traffic. Defining the lists do not have any effect until they are applied to the interfaces.

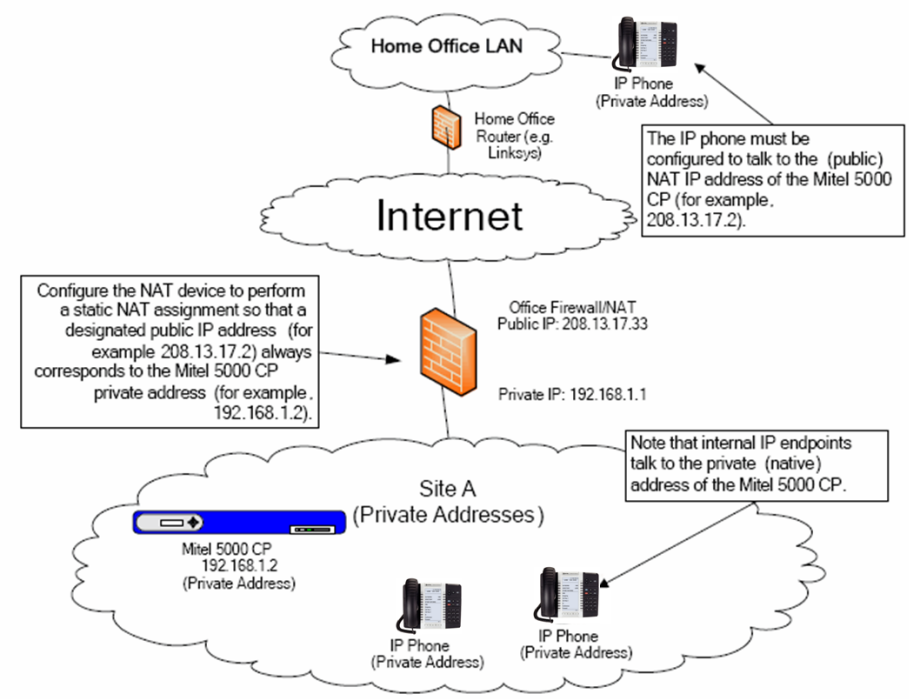
```
ip access-list extended web
  permit ip any  any

ip access-list extended Voice
  permit udp any  host 208.132.23.66 range 6004 6243 log

ip access-list extended ITP
  permit tcp any  host 208.132.23.66 eq 5566  log
  permit udp any  host 208.132.23.66 eq 5567  log

ip access-list extended InterTelNetworking
  permit tcp any  host 208.132.23.66 eq 5570  log
```

- The following commands define the policies for the different interfaces. Each policy can specify more than one access list. In the Public policy-class, voice ports, the ITP (phone) ports, and the Networking ports are allowed.

```
ip policy-class DMZ
  allow list web
!
ip policy-class Private
  allow list self self
  allow list wizard-ics
!
ip policy-class Public
  allow list Voice
  allow list ITP
  allow list InterTelNetworking
```

Private Network With Near-End NAT Traversal

The following diagram shows an example configuration.



The following commands define the VLANs within the switch. This is necessary to be able to route/firewall between the LAN and the Internet.

```
vlan 1
  name "Default"

!
```

- This is a set up of some of the ports in the default VLAN. Repeat for as many ports as you need to configure.

```
interface eth 0/1
  no shutdown
  switchport access vlan 1
!
interface eth 0/2
  no shutdown
  switchport access vlan 1
!
```

- Each VLAN performs as a port on the router with its own IP address. Access policies are also applied to each interface. The access policies are defined as follows:

```
interface vlan 1
  ip address  192.168.1.1  255.255.255.0
  access-policy Private
  no shutdown
```

In this example, the connection is to the Internet with a T1. Set up the IP address and apply the access policy (defined further below). In this example, assume that the ISP is configured with this address as a gateway for the (public) NATed addresses.

```
interface t1 1/1
  clock source line
  tdm-group 1 timeslots 1-24 speed 64
  ip address  208.13.17.33  255.255.255.0

  access-policy Public
  no shutdown
```

- The commands below define access lists for the different traffic types. These defined lists do not have any effect until they are applied to interfaces.
  - The following commands define an access list so that we can communicate with the internal trusted network.

```
ip access-list extended self
  remark Traffic to NetVanta
  permit ip any  any    log
!
```

  - The following commands define the set of communications coming from any internal hosts *except* the MiVoice Office 250.

```
ip access-list standard PrivateHosts
deny host 192.168.1.2
permit any log
```

  - The following commands define the set of communications coming from the MiVoice Office 250.

```
ip access-list standard OutIT5KMain
permit host 192.168.1.2 log
!
```

  - The following commands defines the communications that we want to allow to the MiVoice Office 250. In this example, this is limited to the communications required for ITP phones.

```
ip access-list extended InIT5KMain
  permit tcp any  host 208.13.17.2 eq 5566  log
  permit udp any  host 208.13.17.2 eq 5567  log
  permit udp any  host 208.13.17.2 range 6004 6247   log
```

- These commands define the policy for the interface to the private (trusted) network. Each policy can specify more than one access list. From a security perspective, allow everything from the trusted network. The “nat” commands define the behavior of outbound NAT. If the packet is from

the MiVoice Office 250, translate the source address to the specified public address. The second NAT command uses PAT on IP 208.13.17.33, the address on the public interface, for any other internal node.

```
ip policy-class Private
  allow list self self
nat source list OutIT5KMain address 208.13.17.2 overload
nat source list PrivateHosts address 208.13.17.33 overload
!
```

- The following commands define the policies for the interface to the public (untrusted) network. In the Public policy-class, you are only translating (and therefore allowing) the ports required for ITP (phone) as defined in the access list `InIT5KMain`.

```
ip policy-class Public
  nat destination list InIT5KMain address 192.168.1.2
!
```

Non-NATed DMZ Configuration

The following examples illustrate the configuration for ITP phones and Private Networking.

- The following commands define the VLANs within the switch. This is necessary to be able to route/firewall between the LAN and the DMZ.

```
vlan 1
  name "Default"
vlan 2
  name "DMZ"
!
```

- The following commands set up the Ethernet switch ports for specific VLANs. In this example, take the first two ports and associate them with the DMZ VLAN.

```
interface eth 0/1
  no shutdown
  switchport access vlan 2
!
interface eth 0/2
  no shutdown
  switchport access vlan 2
!
```

Set up some of the ports in the default VLAN. This example shows two ports.

```
interface eth 0/3
  no shutdown
  switchport access vlan 1
!
interface eth 0/4
  no shutdown
  switchport access vlan 1
!
```

Each VLAN performs as a port on the router with its own IP address. Access policies are applied to each interface. The access policies are defined further below.

```
interface vlan 1
  ip address  192.168.1.1  255.255.255.0
  access-policy Private
  no shutdown
interface vlan 2
  ip address  208.132.23.64  255.255.255.192
  no shutdown
!
```

- In this example, the connection is to the Internet with a T1. Set up the IP address and apply the access policy (defined further below).

```
interface t1 1/1
  clock source line
  tdm-group 1 timeslots 1-24 speed 64
  ip address  208.13.17.33  255.255.255.252
  access-policy Public
  no shutdown
```

- The following commands define access lists for the different traffic types. Defining the lists does not have any effect until they are applied to the interfaces.

```
ip access-list extended web
  permit ip any  any

ip access-list extended Voice
  permit udp any  host 208.132.23.66 range 6004 6247 log

ip access-list extended ITP
  permit tcp any  host 208.132.23.66 eq 5566  log
  permit udp any  host 208.132.23.66 eq 5567  log

ip access-list extended InterTelNetworking
  permit tcp any  host 208.132.23.66 eq 5570  log
```

- The following commands define the policies for the different interfaces. Each policy can specify more than one access list. In the Public policy-class, allow the voice ports, the ITP (phone) ports, and the Networking ports.

```
ip policy-class DMZ
  allow list web
!
ip policy-class Private
  allow list self self
  allow list wizard-ics
!
ip policy-class Public
  allow list Voice
  allow list ITP
  allow list InterTelNetworking
```

Appendix C

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- **Lighttpd**: The Web server provided with the MiVoice Office 250 v2.0 and later (see page 475 for details).
- **MD5**: Encryption code provided with the MiVoice Office 250 (see page 476 for details).
- **SQLite:** An embedded SQL database engine provided with the MiVoice Office 250 v5.0 and later (see page 476 for details).

To request open source code modified by Mitel, see “Requests for Source Code” on **page 1-477**.

Abbreviations

This appendix refers to the selected abbreviations listed in the following table.

Table 75: Selected Abbreviations

Abbreviation	Meaning
GNU	GNU’s Not Unix
GPL	General Public License
LGPL	Lesser GPL

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Lighttpd

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Appendix D

Phones

Introduction

This chapter describes the phones and features supported on the MiVoice Office 250. The system supports the following phone types:

- **IP Phones** below
- Digital Telephones on **page 500**
- SIP Phones on **page 508**
- Analog Single Line Phones on **page 509**
- Administrator Phones on **page 511**

IP Phones

The MiVoice Office 250 supports various Internet Protocol (IP) phones. These phones provide the functionality that allows you to make calls through the IP network on the MiVoice Office 250.

Except for the features listed in the bullets below, hard IP phones operate the same as traditional digital telephones installed directly on the system. The following list summarizes the features that are ***not*** supported on hard IP phones.

Hard IP phones:

- Cannot use a PC Data Port Module (PCDPM) or Modem Data Port Module (MDPM).
- Do not support Desktop Open Architecture Interface (OAI) applications.
- Do not have a secondary voice path and cannot support Off-Hook Voice Announce (OHVA).
- Generally do not support Enhanced Speakerphone Mode (except the 8600, 8664/8665/8668 phones).

WARNING

Possible Delay in Local Emergency Response to Remote Sites.

IP phone users should be alerted to the following hazardous situations:

- If an Emergency Call phone number is dialed from an IP phone located at a remote site that is ***not*** equipped with a correctly configured gateway, the call will be placed from the location where system chassis is installed rather than from the location where the emergency call is made.

  In this situation, emergency responders may be dispatched to the wrong location. To minimize the risk of remote site users misdirecting emergency responders, Mitel recommends regular testing of MGCP/SIP gateway trunk(s) for dial tone.

- If uninterruptible power supply (UPS) protection has ***not*** been installed as part of the MiVoice Office 250, IP and SIP phones will ***not*** operate when electrical power fails either at remote sites or at the main system location.

To place calls during a power failure in this situation, IP and SIP phone users can only use a single line phone connected to one of the power failure bypass circuits built into the system chassis. If a phone connected to a power failure bypass circuit is not available, users should make emergency calls **from a local phone *not* connected to the system**. For more information about the Power Bypass feature, refer to the Installation chapter in the *MiVoice Office 250 Installation Manual*.

The IP device interface to the telephone system is an Ethernet IEEE 802.3 100 Mbps unshielded twisted pair (UTP) interface (RJ-45). The hard IP and multi-protocol phones use flash memory and their software can be updated over the Local Area Network (LAN).

Figure 3 through **Figure 21** show IP phones and phone features supported on the MiVoice Office 250.

5212

Figure 3 shows the 5212 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5212/5224 IP Phone User Guide,* part number 550.8122.

Figure 3: 5212

5224

Figure 4 shows the 5224 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5212/5224 IP Phone User Guide,* part number 550.8122.

Figure 4: 5224

5304

Figure 5 shows the 5304 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5304 IP Phone User Guide,* part number 550.8034.

Figure 5: 5304

1 – Liquid Crystal Display (LCD)
2 – Ring/Message Indicator
3 – External speaker
4 – Handset
5 – Programmable buttons
6 – Volume buttons
7 – Dialpad buttons
8 – Hookswitch

5312

Figure 6 shows the 5312 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5312/5324 IP Phone User Guide,* part number 550.8035.

Figure 6: 5312

1 – Handset
2 – Display
3 – Ring/Message Indicator
4 – Feature buttons
5 – Dialpad buttons
6 – Programmable buttons
7 – External speaker

5320

Figure 7 shows the 5320 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5320/5330/5340 IP Phone User Guide,* part number 550.8123.

Figure 7: 5320

1 – Handset
2 – 11 Programmable buttons[a]
3 – Display[b]
4 – Ring/Message Indicator
5 – Feature buttons
6 – Dialpad buttons
7 – Navigation page buttons
8 – External speaker

a. The top three buttons on the 5320 IP Phone do not have LEDs. The 5320 supports one page of programmable keys. The 5320 IP Phone has 11 programmable buttons. The 5320 supports one page of programmable keys.

b. The 5320 IP Phone does not have a backlit display.

5324

Figure 8 shows the 5324 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5312/5324 IP Phone User Guide,* part number 550.8035.

Figure 8: 5324

1 – Handset
2 – Display
3 – Display buttons
4 – Ring/Message Indicator
5 – Feature buttons
6 – Dialpad buttons
7 – Programmable buttons
8 – External speaker

5330

Figure 9 shows the 5330 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5320/5330/5340 IP Phone User Guide,* part number 550.8123.

Figure 9: 5330

5340

Figure 10 shows the 5340 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5330/5340 IP Phone User Guide,* part number 550.8123.

Figure 10: 5340

5360

Figure 11 shows the 5360 IP Phone and features. For more information about the phone and system features, refer to the *Mitel 5360 IP Phone User Guide,* part number 550.8124.

Figure 11: 5360

5610

Figure 12 shows the Mitel 5610 Cordless Handset and IP DECT (Digital Enhanced Cordless Telecommunications) Stand. For instructions on using the 5610 Cordless Handset, refer to the MiVoice Office 250 version of the 5610 IP DECT Cordless Handset Quick Reference Guide (part number 57011425).

Figure 12: 5610 and IP DECT Stand

8600

Figure 13 shows the 8600 Multi-Protocol Phone and features. For more information about the phone and system features, refer to the *Mitel 8600 Phone User Guide,* part number 550.8115.

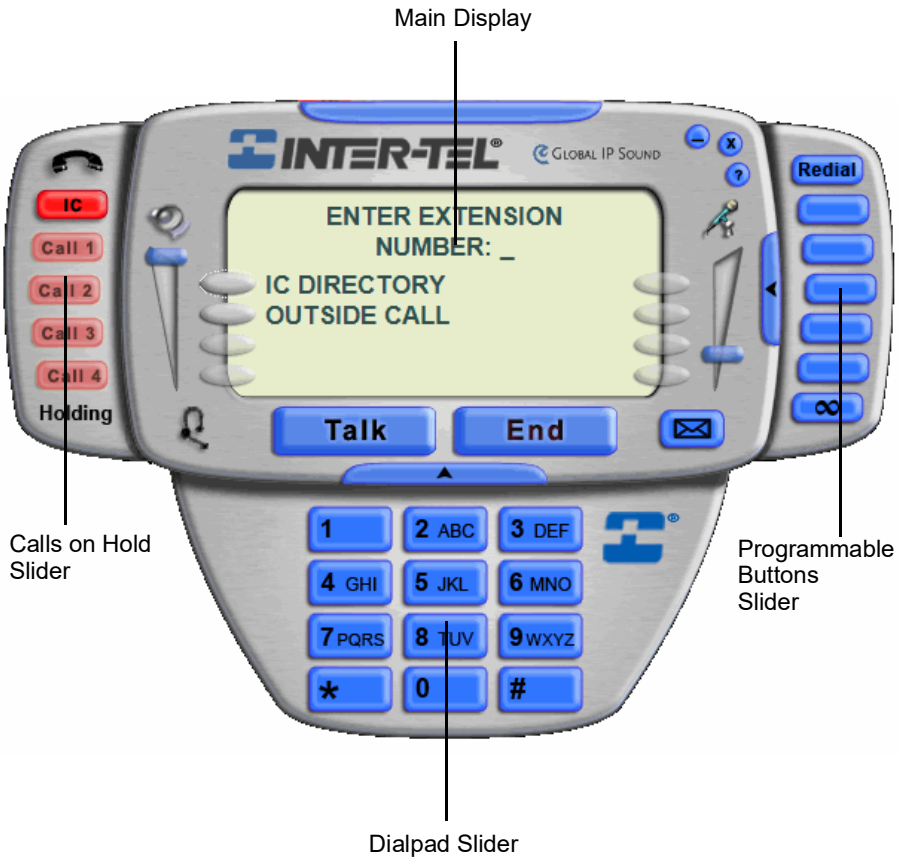
> **NOTE** 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

Figure 13: 8600

8602

Figure 14 shows the 8602 SoftPhone. For more information, refer to the *8602 User Guide,* part number 835.3109.

Figure 14: 8602



8620/8622

Figure 15 shows the 8620/8622 Multi-Protocol Phone and features. For more information about the phone and system features, refer to the *Mitel 8620/8622 Phone User Guide,* part number 550.8119.

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

Figure 15: 8620/8622

NOTE The 8622 phone has two Ethernet ports on the bottom of the phone. The 8620 phone has one port.

8660

Figure 16 shows the 8660 IP Phone and features. For more information about the phone and system features, refer to the Mitel 8560/8660 Phone User Guide, part number 550.8113.

Figure 16: 8660

1 – Handset
2 – Display
3 – Message Indicator lamp
4 – Menu buttons
5 – Programmable buttons
6 – Dialpad buttons
7 – Feature buttons
8 – Speaker

8662

Figure 17 shows the 8662 Multi-Protocol Phone and features. For more information about the phone and system features, refer to the Mitel 8662 Phone User Guide, part number 550.8117.

NOTE 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

Figure 17: 8662

NOTE The 8662 phone has one Ethernet ports on the bottom of the phone.

8664/8665/8668

Figures **18**, **19**, and **20** show the 8664, 8665, and 8668 wireless IP phones. For more information about the phones and system features, refer to the *8664/8665/8668 User Guide,* part number 935.4522.

Figure 18: 8664

Figure 19: 8665

Figure 20: 8668

8690

Figure 21 shows the 8690 Multi-Protocol Phone and features. For more information about the phone and system features, refer to the *8690 Phone User Guide,* part number 550.8116.

> **NOTE** 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

Figure 21: 8690

Digital Telephones

The following pages show digital telephones and phone features supported on the MiVoice Office 250.

NOTES
The Professional Display, Associate Display, and Basic digital telephone are supported by U.S. software only.

85xx-series digital telephones do not support full-duplex operation.

8500

The 8500 Digital Telephone external form factor is the same as the 8600 IP Phone (see page 491). For more information about the phone and system features, refer to the *Mitel 8500 Digital Telephone User Guide,* part number 550.8114.

8520

The 8520 Digital Telephone external form factor is the same as the 8620/8622 IP Phone (see page 493). For more information about the phone and system features, refer to the *Mitel 8520 Digital Telephone User Guide,* part number 550.8112.

8524/8525/8526

Figure 22 shows the 8524/8525/8526 wireless digital telephones. For more information about the phones and system features, refer to the *8524/8525/8526 User Guide,* part number 935.4529.

Figure 22: 8524/8525/8526

8528

Figure 23 shows the 8528 Digital Telephone and features. For more information about the phone and system features, refer to the *Mitel 8528 Digital Telephone User Guide,* part number 550.8032.

Figure 23: 8528

1 – Handset
2 – Display
3 – Ring/Message Indicator
4 – Feature buttons
5 – Dialpad buttons
6 – Programmable buttons
7 – External speaker

8560

The 8560 Digital Telephone external form factor is the same as the 8660/8662 IP Phone (see page 494). For more information about the phone and system features, refer to the *Mitel 8560/8660 Phone User Guide,* part number 550.8113.

8568

Figure 24 shows the 8568 Digital Telephone and features. For more information about the phone and system features, refer to the *Mitel 8568 Telephone User Guide,* part number 550.8033.

Figure 24: 8568

1 – Handset
2 – Display
3 – Menu buttons
4 – Ring/Message Indicator
5 – Feature buttons
6 – Dialpad buttons
7 – Programmable buttons
8 – External speaker

Associate

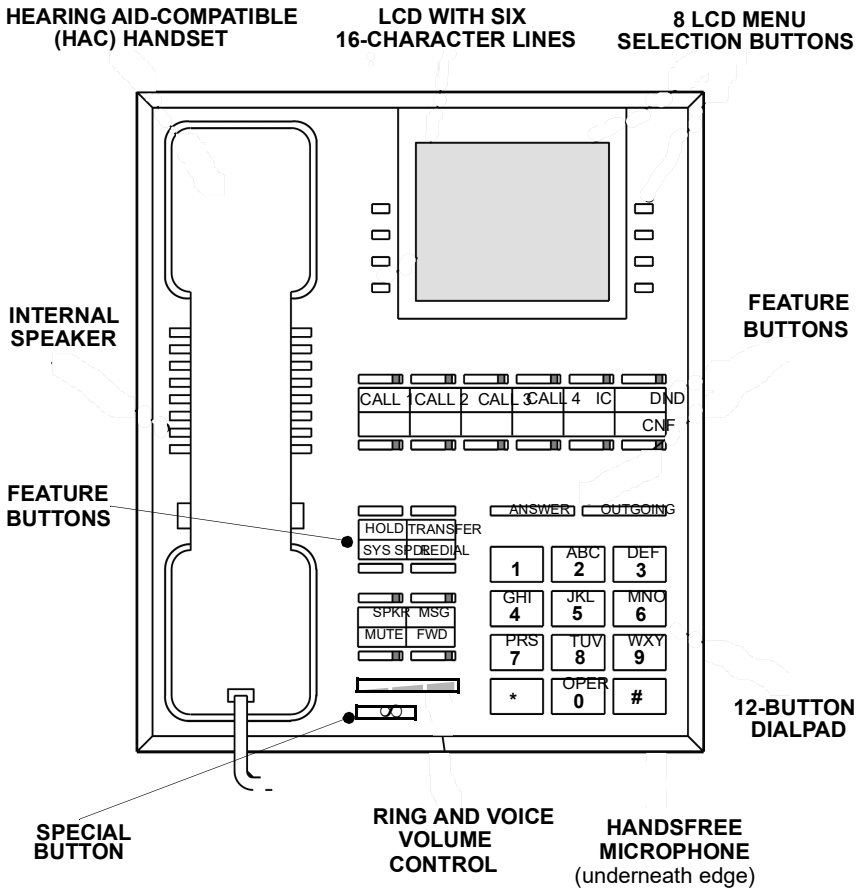
Figure 25 shows the Associate Display Digital Telephone and features. For more information about the phone and system features, refer to the *Associate Display and Basic Digital Telephone User Guide,* part number 560.8003.

Figure 25: Associate

Executive

Figure 26 shows the Executive Display Digital Telephone and features. For more information about the phone and system features, refer to the *Executive Display and IP PhonePlus User Guide,* part number 550.8108.

Figure 26: Executive



Professional

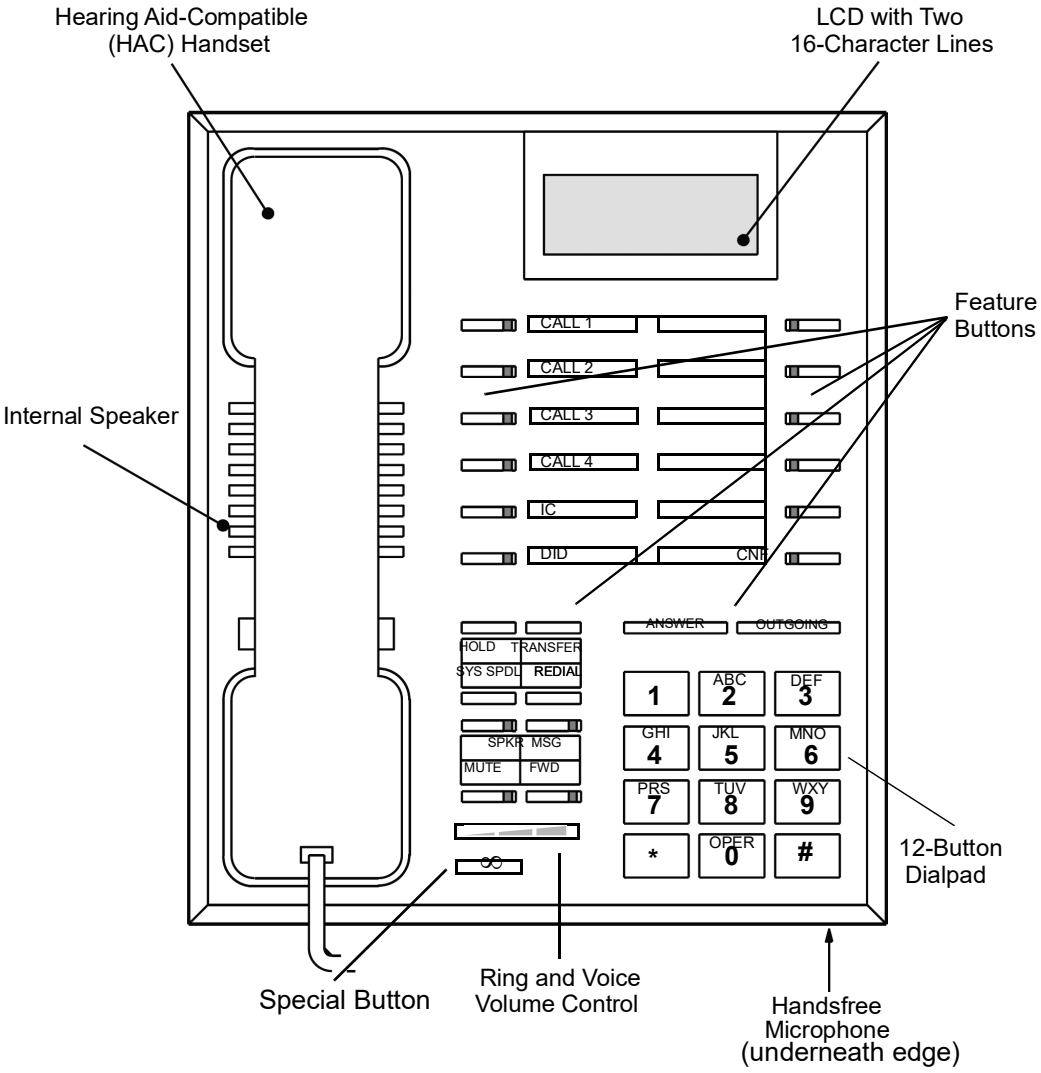
Figure 27 shows the Professional Display Digital Telephone and features. For more information about the phone and system features, refer to the *Professional Display and IP PhonePlus User Guide,* part number 560.8002.

Figure 27: Professional

Standard

Figure 28 shows an illustration of the Standard Display Digital Telephone and features. For more information about the phone and system features, refer to the *Standard Display and Basic Digital Telephone User Guide,* part number 550.8109.

Figure 28: Standard/Basic



SIP Phones

> **NOTE** 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

To change the modes on 8600, 8620/8622, and 8662 multi-protocol phones, do one of the following procedures:

- **Dialpad**: To switch to ITP mode, hold down **5** and **6** on the dialpad while powering up.

> **NOTE** The preferred method for changing between SIP mode and ITP mode is through the Administrative Web Portal (AWP)—see page 534.

- **Web Interface**: Change the operation mode through the Web interface (see **page 534**).
- **TFTP Server**: Change the configuration files using the TFTP server. For more information, refer to the “Installation” chapter in the *MiVoice Office 250 Installation Manual*.

To change the modes on the 8690 multi-protocol phone:

Change the operation mode through the Web interface (see **page 534**).

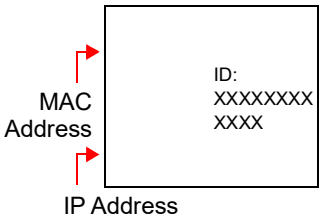
ITP Mode

ITP mode phones connect to the system through a TCP/IP stream to the IP Resource Application (IPRA) running a Mitel-proprietary protocol. This is the same protocol that the 8660 uses, which is a modified version of the protocol that the 8500 series digital telephones use. The main difference is that the multi-protocol phones connect to the system through IP.

The advantage of ITP mode over SIP mode is that ITP mode allows you to use all of the system features and functionality that the system provides for a traditional phone. ITP mode supports most system features. The disadvantage of ITP mode over SIP mode is that when the phone is in ITP mode, it cannot connect to another vender switch. All of the multi-protocol phones are configured to use ITP by default.

To support multi-protocol phones in ITP mode, a Call Processing software license is required.

Phone IP and MAC Address in ITP Mode: Unlike phones in SIP mode, a phone in ITP mode does not support the Show IP feature code (300) to display the phone IP address. V2.0.06 or later phone firmware allows a phone in ITP mode to display the IP address and MAC address on the LCD, as shown at right.



To display the IP and MAC addresses:

Press the Special button twice followed promptly by the **MUTE** button.

Analog Single Line Phones

Analog single line phones are connected to Single Line Modules (SLM-4 or SLM-8), to IP Single-Line Adapter (SLA) circuits, or to SLA circuits, which in turn are connected to available Digital Endpoint Module (DEM-16) circuits. Also, one or two single line phones can be connected to the built-in ports on the back of the Base Server. OPXs can be connected to SLAs or T1M or T1M-2 modules.

- Each IP SLA can support one or two single line phones or DID trunks.
- T1M or T1M-2 module circuits can be used for installing off-premises single line phones.

Single line phones incorporate the following features:

- Audio message waiting indication capability
- Programmable hookflash/recall recognition
- Programmable ring cadence and gain levels for OPX applications
- Optional SLA for connecting an off-premises extension to the DEM-16
- Caller ID and Calling Line Identification [CLID in Europe] support
- Pulse dialing support for single line phones

Feature Access Using Single Line Phones

Single line phone users gain access to the system software features by hookflashing [recalling, in Europe] and entering DTMF-transmitted feature codes. Hookflashing [Recalling] is the action of pressing and quickly releasing the hookswitch.

Ring Zones and Extended Ring Cadences

In some installations, where more than one single line device is connected in series to an SLM or DEM-16 circuit, zoned ringing may be necessary. The system can support devices with up to 3 Ringer Equivalence Numbers (REN) per circuit. However, the system can only ring 8 RENs per module. If the installation exceeds eight RENs on any SLM or DEM-16, the system-wide SLM Ring Zones flag must be enabled. Also, all circuits on the SLM or DEM-16 should have the Extended Ring Cadences phone flag enabled to allow the longer cadence to be used instead of the short internal cadence.

For more information about system flags, refer to the “System Settings” chapter in the *MiVoice Office 250 Features and Programming Guide*.

NOTE IP SLAs, SLAs, and T1M and T1M-2 module circuits *cannot* support message waiting lamps.

Table 76 summarizes the differences between standard and extended ring cadences for the various types of single line phones used on the Mitel system in the United States.

Table 76: Standard and Extended Ring Cadences for Single Line Phones

Single Line Device	Cadence Type	Description
SLA or T1 OPX	Standard IC Ring	This is a repeating 2-second cadence that rings twice. Each ring lasts 200 ms and is separated from the next ring by 400 ms of silence. The remaining 1.2 seconds is silent.
	Extended IC Ring	This is a repeating 6-second cadence that rings twice. Each ring lasts 800 ms and is separated from the next ring by 400 ms of silence. The remaining 4 seconds are silent
	Standard CO Ring	This is a repeating 6-second cadence that rings once for 2 seconds followed by 4 seconds of silence.
	Extended CO Ring	Same as Standard CO Ring cadence.

Off Premises Extensions (OPX)

Off premises extensions are remote single line DTMF phones connected to the system through telephone company OPX circuits or customer-provided circuits.

OPX phone users gain access to the features by hookflashing [recalling] and entering feature codes. Hookflashing [Recalling] means to press and quickly release the hookswitch.

Some telephone company OPX networks do not recognize certain single line phone ring cadences used in the system. This incompatibility results in an inability to provide certain ring signals to OPX. Because of this, the system is equipped with a phone option (Extended Ring Cadences) that allows the ring cadences be altered for individual single line circuits.

Another programming option, the OPX Gain Levels flag, allows the receive and transmit gains to be automatically adjusted to improve OPX communication with the CO.

OPX circuits connect to IP SLA devices across T1/E1 spans.

IP Single-Line Adaptor

The IP Single-Line Adaptor (SLA) provides interface between a single Local Area Network (LAN) port and one single line analog phones. For installation instructions, refer to the “Installation” chapter in the *MiVoice Office 250 Installation Manual*.

Administrator Phones

Any phone, attendant or non-attendant, can be assigned for administrator use through DB Programming. However, only display phones should be designated for administrators so they can see system alarms and to make programming easier. For administrator phone feature descriptions and instructions, refer to the following guides:

- *MiVoice Office 250 Phone Administrator Guide*, part number 580.8001
- *MiVoice Office 250 Unified Voice Messaging Administrator Guide*, part number 580.8009.

Administrator phones can be used to:

- Place the local node or other nodes in the network in night mode or day mode.
- Set the date and time of the local node.
- Set the network date and time and re-synchronize clocks in the network.
- Make database changes.
- Program System Speed Dial numbers on the local node.
- Receive and clear displayed system and network alarms.
- Create Custom Audiotex Recordings
- Change phone Extension Numbers
- Swap phone extension numbers
- Change default Do Not Disturb Messages
- Change default Reminder Messages
- Program selected trunk options
- Use diagnostic mode features to:
  - Freeze and unfreeze database history on local nodes or other nodes in the system.
  - Print error logs.
  - Seize specific devices for troubleshooting purposes.

Phone Physical Features

The following sections describe physical features of phones that operate on the MiVoice Office 250.

DSS/BLF Unit

The MiVoice Office 250 supports two 60-button Direct Station Select/Busy Lamp Field (DSS/BLF) Units and one 50-button model. Although you can use either type of DSS/BLF Unit on any of the phones, the type of phone determines which is the more appropriate unit to use.

- PC Data Port Module (PCDPM)-equipped Executive, Professional, Standard, and Associate Display phones can support up to four 60-button DSS/BLF units. Up to twenty 60-key DSS/BLF Units can be installed on a system. The 6-rows-by-10-columns model, is designed for Executive and Standard Display phones, and the 12-rows-by-5-columns model is designed for Professional and Associate Display phones. For more information, refer to the “Specifications” chapter in the *MiVoice Office 250 Installation Manual*.
- PCDPM-equipped 8520 and 8560 phones support up to four 8450 50-button DSS/BLF units. Up to 20 of the 5-rows-by-10-columns DSS/BLF units can be installed on a system. For specifications and installation instructions, refer to the “Specifications” and “Installation” chapters in the *MiVoice Office 250 Installation Manual*.

DSS/BLF Units always require dedicated AC transformers. For a diagram of AC transformer requirements or unit installation instructions, refer to the “Specifications” and “Installation” chapters in the *MiVoice Office 250 Installation Manual*

Each DSS/BLF unit provides one-button access to 50 or 60 extension numbers, trunks, features, or other resources, and four units provide access to 200 or 240 extensions, features, and resources. Numbers programmed for the buttons can be phone extension numbers, hunt group numbers, trunk access codes, or feature codes.

The layout of the buttons and the numbers are assigned using one of 20 keymaps in DB Programming. The intercom number/button assignments of each keymap are programmed in four lists, pages 1–4. For example, single DSS/BLF Units are assigned page 1 of the desired keymap, tandem units are assigned two pages of the desired keymap—page 1 for the first unit and page 2 for the second unit, and so on.

For more information about keymaps, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Together, the lamps in the buttons create a Busy Lamp Field that indicate the status of each station, hunt group, or other feature/resource assigned to the buttons. If assigned to a station, the LED indicator in the button is solidly lit when the associated station is busy or unplugged, flashes slowly when the station is in Do-Not-Disturb (DND), flashes rapidly when the station has a call ringing in, or flutters continuously if the station is causing a STATION OFF-HOOK system alarm. If assigned to a hunt group, the LED indicator is solidly lit when all stations in the hunt group are unavailable—whether busy, in DND, or removed from the hunt group. The LED indicator flashes rapidly when a call is camped-on to a hunt group. If assigned to a feature or resource, the LED indicator is lit to show busy status and unlit to show idle status.

NOTE Group Call Pick-Up does not affect DSS lamp status any differently than standard hunt group calls or individual calls.

Speakerphones

NOTE The network allows handsfree intercom calls when calling from one node to another, unless the call camps on to the node before being sent to the other node.

Executive Display and Professional Display Speakerphone

Each Executive Display and Professional Display phone has a built-in, integrated speakerphone that allows on-hook dialing and handsfree operation on outside calls and inside (intercom) calls. The Executive Display and Professional Display phone speakerphone can be used in standard mode or in enhanced mode. See “Enhanced Speakerphones for Digital Telephones” on page 513.

The Executive Display and Professional Display phone speaker can provide background music if an external music source is installed. As well, the phone speaker can be used to receive paging messages.

A programming option based on the speakerphone type can be used to disable the phone’s speakerphone capabilities. Or, the user can disable the speakerphone for intercom calls using the Handsfree On/Off and/or Ring Intercom Always On/Off feature codes.

NOTE On certain handsfree-to-handsfree intercom calls, voice volume levels may cause feedback to occur. If this happens, one phone user should pick up the handset.

Other Digital Telephone Speakerphones

Standard and Associate Display, and digital telephones have built-in speakers and microphones that allow the user to answer intercom calls handsfree, dial while on-hook, monitor calls, and listen to background music.

These phones can be programmed in individual phone programming to allow the use of system resources for standard and enhanced speakerphones. See “Enhanced Speakerphones for Digital Telephones” on page 513. To provide standard and enhanced speakerphone resources for Standard and Associate Display, and digital telephones, the system must be equipped with a DEI and at least one DEM-16.

Each time a speaker/microphone phone user presses the Speaker button while on-hook, the standard switched speakerphone is activated, and the DSP resource is assigned to that phone for the length of the call. If a phone user requests the speakerphone function when speakerphone resources are not available, the user hears reorder tones, and the display shows NO SPKRPHN AVAIL STAY OFF HOOK. If the user attempts to change to the speakerphone and hangs up during a call when speakerphones are unavailable, the call will be forced into a monitor-only mode. The user hears the other party but is not allowed to transmit to them. In this mode, the user’s **MUTE** button is lit, and the display shows NO SPKRPHN AVAIL COME OFF HOOK. Pressing the lit **MUTE** button results in a burst of reorder tone, and you must lift the handset to speak.

Enhanced Speakerphones for Digital Telephones

Enhanced speakerphone mode allows for full-duplex operation over a limited volume range. The range depends on trunk characteristics; typically, the range is 1–3.

To use an enhanced speakerphone on a digital telephone, the user establishes a call and presses the **Special** button and then the Speaker button. Or, the user presses the **Special** button and enters the Enhanced Speakerphone feature code, 310. About one second of white noise is heard at this point. Both parties should avoid speaking during this white noise because it is used to calibrate the echo-canceling circuitry. After the white noise, a confirmation tone is heard. On Executive Display and Professional Display phones, users hear the tone, but the other party does not. On Standard and Associate Display phones, and digital telephones, both the user and the other party hear the tone. The user is then connected in an enhanced handsfree call. If, due to trunk conditions, the speakerphone cannot be calibrated correctly, the enhanced speakerphone is not enabled and the display shows CANNOT ENABLE ENHANCED MODE.

NOTES

In general, IP phones and non-IP phones that connect to IP calls—those that are routed through the IPRA—***cannot*** enable enhanced speakerphone mode. However, sometimes the system may not detect that a call is connected over an IP network. If this occurs, the system may allow users to enable enhanced speakerphone mode, which will affect the call quality.

It is normal for an outside party to hear a minimal amount of his or her own echo when speaking to a digital telephone in enhanced speakerphone mode. This is usually acceptable to most callers on local calls and on many long-distance calls.

However, the amount of echo may be objectionable on some calls. An unacceptable amount of echo can occur on calls established using IP Networking or IPRA and some T1 and long-distance calls involving significant delays—for example, cross-country T1 calls, overseas calls, or calls via satellite. For these types of long-distance calls, including ones connected through IP Networking, the enhanced speakerphone mode should *not* be used.

If an enhanced speakerphone call is placed on hold and then taken off hold on an Executive and Professional Display phone, the call remains an enhanced speakerphone call. However, if a call is placed on hold at a Standard or Associate Display phone, and digital telephone, the call returns to standard speakerphone mode.

A station can support only one enhanced speakerphone call. For example, if the user of an Executive or Professional Display phone makes an enhanced speakerphone call, puts it on hold, and establishes another enhanced speakerphone call, the first call will be converted to a standard speakerphone call.

A digital telephone user can set up a conference and then, once it is established, press ∞ and the Speaker button (or press ∞ and enter the Enhanced Speakerphone feature code – 310) to have an enhanced speakerphone call in the conference.

NOTE

On certain handsfree-to-handsfree intercom calls, voice volume levels may cause feedback to occur. If this happens, one station user should pick up the handset. Two display phones cannot be in enhanced speakerphone mode while on a call together. One phone must be off-hook to allow the other phone to enter enhanced speakerphone mode. If both phones are on-hook, there is no benefit to having one phone in enhanced mode.

Speakerphone Guidelines

Keep in mind the following guidelines when using the standard mode:

- Talk within four feet of your phone.
- Keep your phone at least four feet away from noise generators (e.g., PC fans). Speakerphones perform better in a quiet office environment.

- Use any angle your 8500-series phone supports as the angle makes very little difference to levels and performance even though the microphone is “hidden” when vertical.
- Avoid using the speakerphone as a “conference room” speakerphone. Conference room speakerphones, such as a Polycom®, typically have as many as three to five microphones that stretch 10 feet or more end-to-end to pick up the voices of all people in the room. A phone has just one microphone and works best when you and or other parties are within four feet of the phone. If you are in a conference room where there are multiple parties farther than four feet from the phone, it is best to use a stand-alone speakerphone designed specifically for that purpose.
- Avoid using the standard mode when the far-end party is in a “noisy” environment — typical with cell phone callers.
- Avoid using the standard mode when the call is a conference call with multiple parties because the likelihood of one or more of the parties having background noise is high.
- Standard speakerphone operation is half-duplex or “switched” and switches to the “loudest” party. Half-duplex means that if both parties talk at the same time, only one party—the loudest one—will be heard.
- The far-end talk “level” is based on your speaker volume setting (1–8). The louder you set the volume, the more difficult it becomes to interrupt the far-end party.
- As a general rule, the far-end talk level is “louder” than the near-end talk level giving the far-end party priority when talking.
- The farther away you are from the phone, the lower the talk level.
- The speakerphone adjusts to ambient “noise” level in the room, which is why you need to keep the phone away from noise generators such as PC fans and talk within four feet of your speakerphone.

Keep in mind the following guidelines when using the enhanced mode:

- Follow the same guidelines as those listed for the “standard” speakerphone, and, in addition to the guidelines for an enhanced speakerphone.
- Inform the far-end party about the burst of white noise before you enter the enhanced mode.
- Use volume levels 1–3 for full-duplex operation.
- Avoid using the enhanced mode on calls with significant delay.
- Avoid using the enhanced mode on volume levels 4–8 since the speakerphone transitions back to half-duplex operation.
- The enhanced speakerphone provides full-duplex operation at lower volume settings (1–3) but transitions to half-duplex operation at higher volume settings (4–8). Full-duplex means both parties can talk at the same time and hear each other.
- The enhanced mode does ***not*** perform acoustic echo cancellation, so the far-end party will always hear their own acoustic echo.
- The far-end party will hear all near-end background noises.
- The far-end party must be on a handset for full-duplex operation.

Full Duplex Speakerphones for IP Phones

For full-duplex operation without any limitations, use one of the following Mitel IP phones:

- 5224
- 5312
- 5320
- 5324
- 5330
- 5340
- 5360

See “IP Phones” on page 480 for more details about these phones.

LCD

The following sections provide information about the phone liquid crystal display (LCD).

Menu Keys

Some phones have feature-related menus that change with the function being performed by the phone (intercom call, outgoing call, programming, etc.). Menu selection keys that are used for menu features appear next to or below the CALL buttons.

All Executive Display, Professional Display, and 8560 phones have a six-line display. Each line has 16 characters. The top two LCD lines display standard call information and show the same displays as other display phones. The remaining four LCD lines display menu options. These lines have menu selection buttons on both sides (eight buttons) that are used for executing menu features. When two prompts appear on an LCD line, such as PREVIOUS and NEXT, the button next to the word is the one that selects named feature. When only one prompt appears on a display line, such as DO-NOT-DISTURB, then pressing the button on either side of the prompt selects the named feature.

A system-wide flag called “Alternate Keyset Menu Displays” determines the display that 8660 phone users see during an outgoing call. In the default state, the display shows the menu (shown) with Transfer-To-Hold on the first menu and Flash on the “More...” menu.

HOLD	SYS HOLD
TRANSFER	CNF
TRANSFER-TO-HOLD	
MUTE	MORE...

If the Alternate Keyset Menu Displays flag is enabled, 8660 phone users see the menu with Flash (shown) on the first menu and Transfer-To-Hold in the “More...” menu.

HOLD	SYS HOLD
TRANSFER	CNF
FLASH	
MUTE	MORE...

Enabling the Alternate Keyset Menu flag allows Centrex features to be accessed more easily, using the FLASH menu option.

Extension Number, Username, Time and Date Display

When a display phone is idle, the extension number, username, time, and date are displayed. While on a call, in DND, or unconditionally forwarded, other displays are shown. In these circumstances, use the following procedure to re-display the date, time, user name, and extension number for 5 seconds.

To display the date, time, username, and extension number:

Press the **Special** button and enter the Display Time/Date feature code (300). The date, time, username, and extension number are displayed for 5 seconds.

Node and IP Address on Power-Up

To aid in installation and troubleshooting procedures, display phones show their phone identification for 5 seconds when power is turned on, when the system is reset, or when the phone line cord is first plugged in or when it is removed and replaced. The extension number and assigned user name appear on the top line of the display and the node number and IP address appear on the bottom line. After the 5-second identification display, the phone returns to the appropriate display, depending on its current status.

All Transient Displays Phone Flag

The “All Transient Displays” phone flag, when enabled, allows all transient call handling displays to appear on the phone display. If it is disabled, the CALL TRANSFERRED TO XXXX and MESSAGE LEFT FOR XXXX transient displays do not appear.

For more information about phone flags, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Each transient display is shown for 5 seconds before another display can take its place. Disabling the flag allows high-traffic phones such as attendants to change displays more quickly after a message is left by the phone user or after a call is transferred.

This option cannot be programmed by the user. It is a database-programmable flag. It is not used for single line phones.

Transient Call Indication on Call Answer Phone Flag

This programmable flag determines if the phone user sees a call display when answering a call by pressing a secondary extension button or by reverse transferring. If enabled, the display indicates if the call was ringing, recalling, transferred, or holding at the other phone.

Feature Keys

The following sections provide information on the phone feature keys.

Hot Keypad Keys vs. SPCL Key Required for Feature Code Entry Flag

Unless programmed to use the Special button when entering a feature code, all phones have “hot” keypad buttons that allow dialing of feature codes, extension numbers, and trunk access codes without pressing the Special button or lifting the handset. As soon as a keypad digit is pressed, the digit is recognized by the system.

There is a system-wide option called “SPCL Key Required For Feature Code Entry” that, if enabled, requires phone users to always press the Special button before entering a feature code.

Regardless of programming, if a phone user presses the Special button while on-hook, the display shows ENTER FEATURE CODE. The user can enter the desired code or press a menu button.

Keymaps

A keymap group determines the layout of the buttons for all the phones assigned to that group. If any of the buttons in the keymap group are user-programmable or undefined, they can be changed by the phone user. Each node contains its own keymap programming.

For more information about keymaps, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Each phone is assigned a standard keymap and can also have an alternate keymap. See page 520 for information about switching between standard and alternate keymaps.

When the system is in the default state, all phones are assigned to the default Keymap 1. There are no individual pre-programmed keymaps for each of the specific phone types. Keymap #1 also acts as the template for new keymaps, regardless of the phone type. For example, if keymap #1 is an 8660 keymap with the default Hold button programmed as a Forward button, any newly created keymaps that have a Hold button, even Standard Display keymaps, will reflect this change. Because keymap #1 is used as a template, it is recommended that you do not change this keymap.

As part of the installation procedure, the installer must make sure the phone buttons are labeled to match their functions.

The buttons in each keymap can be assigned the following functions:

- **Cancel Key:** No further programming is necessary. This button will be used to cancel a feature or cancel an entry when in numeric mode.
- **Call Key:** Up to 10 CALL buttons can be assigned in each keymap. These buttons are used for call access. They are not associated with specific trunks. Instead, they are assigned to calls in the order that the calls are placed or received. In the following example, the user has three CALL buttons.

Table 77: CALL Key Examples

Action	CALL Key 1	CALL Key 2	CALL Key 3
User places call #1	Shows status of call #1		
Call #2 rings in while call #1 is active	Shows status of call #1	Shows ring flash for call #2	
Calls #1 and #2 on hold, call #3 placed	Shows hold flash for call #1	Shows hold flash for call #2	Shows status of call #3

If there are more calls in progress than there are CALL buttons (for example, if another call rings in to the phone in the example), the new call will not have an available call button and will camp on to the phone until a CALL button becomes available; it will then appear under the available button. Pressing an idle CALL button will enter the programmed Outgoing Call feature for that phone. If there is no **IC** button programmed in the keymap, intercom calls will appear under CALL buttons.

NOTE All keymaps should have at least one CALL button if there is not an individual trunk button for every trunk.

- **Feature Key:** These are programmed with feature codes. The feature code appears in the same button location on all phones assigned to the keymap and cannot be changed by the phone user.
- **Forward Key:** The forward button can be programmed with any of the Call Forwarding feature codes. The forwarding code used by the button can be changed by the phone user.
- **Hunt Group Key:** This button dials a hunt group pilot number when pressed and, if assigned to a button with a lamp, will show the status of that hunt group.
- **Individual Trunk Key:** This button is associated with one trunk. It can be used for placing outgoing calls, and for picking up calls on hold and answering incoming calls on that trunk.
- **Intercom (IC) Key:** The **IC** button allows access to intercom channels for placing and answering calls. If an **IC** button is not assigned, an intercom call will appear under a CALL button.
- **Page Zone Key:** A page zone button is programmed with one of the page access codes (9600–9609 or 9600–9649 depending on the software version) for quick access to page zones (0–9 and 0–49 respectively).
- **Programmable Key:** Up to 35 user-programmable buttons can be designated and given default values in the keymap database. They can then be reprogrammed to enter any of the trunk group access codes, extension numbers, or feature codes by the individual phone users as desired. Or, they can be programmed as System or Station Speed Dial bins, or as secondary extension buttons (see below). The phone user or system administrator can return the buttons to default values at any time.
- **Secondary Extension Key:** This type of button allows a phone user to have one-button access to calls ringing or holding at another phone. It should be assigned to a button with a lamp so that the secondary extension button can show the call status. If desired, a phone can have several secondary extension buttons, each associated with a different phone.
- **SPKR Key:** This button is used for turning the phone speaker or headset on and off.

- **Speed Dial Keys:** Each keymap can have up to 10 Speed Dial buttons. Each Speed Dial button can be programmed by the phone user to dial one digit string. If assigned to a button with a lamp, the lamp will show the status of an extension number if it is associated with that button.
- **Trunk Group Key:** These buttons are associated with specific trunk group access codes and are used for placing outgoing calls on trunks in the selected trunk group. If assigned to a button with a lamp, the lamp will be lit if all trunks in the trunk group are in use.
- **Undefined Key:** Any button can be left undefined if it will not be needed. It can be programmed, like a user-programmable button, as described above. If the phone user or system administrator returns the phone to default values, the button will return to being “undefined.”

Phones and Button Locations

The phones have additional buttons and lamps. For phone keymap assignments, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Standard/Alternate Keymap Switching

In database programming, each phone is assigned a standard keymap and can also have an alternate keymap. The user can switch between the keymaps by entering a feature code. The maps can also be switched in the database using the Alternate Keymap phone flag.

For more information about keymaps, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Having two keymaps provides the user with access to more features or trunks. In most situations, the user would have one button programmed with the Switch Keymap feature code (399) in each keymap so that it would be easy to change the keymap. If the user-programmed button has a lamp, the lamp is lit when the alternate keymap is active.

When keymaps are switched, any user-programmable buttons in the maps retain their user-programmed values; they are not returned to default during the switch. However, if the keymaps do not have the same user-programmable buttons in the same positions, the location of individual buttons may change when the keymaps change.

If a user switches keymaps during an outside call, the call will appear on the appropriate trunk or CALL button for that keymap. The button assignment may not be the same for both keymaps. For example, it could appear under a CALL button on one map and an individual trunk button on the other.

If a button is not available in the currently active keymap when a call rings in, the user will still see the correct display, hear ringing, and can answer as usual.

To switch between keymaps when the phone is idle:

While on-hook, enter the Switch Keymap feature code (399). The display shows STANDARD KEYMAP IS ACTIVE or ALTERNATE KEYMAP IS ACTIVE.

To switch between keymaps during a call:

While on a call, press the Special button, and then enter the Switch Keymap feature code (399). The display momentarily shows STANDARD KEYMAP IS ACTIVE or ALTERNATE KEYMAP IS ACTIVE.

Message Lamp

85xx and 86xx phones have a Message Lamp located on the upper, right-hand corner of the phone. This lamp is programmable and is defaulted to the Message feature code (365). If you change the feature code associated with this lamp, it will flash when the feature is active. For example, if you program the lamp for Individual Hold (336), the lamp will flash when a call is held at the phone.

Professional Display, Associate Display, and Basic digital telephones also have a Message Lamp, but it is not programmable. Instead, it flashes whenever the **MSG** button flashes.

Speaker Buttons

On Executive, Professional, Standard, and Associate Display phones, and on digital telephones, handsfree mode is controlled by a Speaker (**SPKR**) button. On 8500, 8520, and 8560 phones, an illustration of a speaker (🔊) represents the Speaker button.

Hold Buttons

On Executive, Professional, Standard, and Associate Display phones and on digital telephones, the Hold feature is controlled by a Hold (**HOLD**) button. On 8500, 8520, and 8560 phones, an illustration of a phone with the handset off-hook (📞) represents the Hold button.

Default Feature Buttons

When the system is in the default state, all phones are in the same keymap group and have the feature buttons. Refer to the appropriate user guide for default feature button functions.

Secondary Extension Appearances

The Secondary Extension Appearances feature allows calls ringing or on individual hold at one “primary” phone to appear on a “secondary extension key” at another phone. For example, a secretary or assistant can have a secondary extension button for a supervisor’s phone, or one phone can have several secondary extension buttons, each associated with a different primary phone. The primary phone and the secondary phone must reside on the same node.

A secondary extension button functions as follows:

- When a call is ringing or holding on any CALL button at the primary phone, pressing the secondary extension button will answer the call. The secondary extension can answer recalls, transfers, callbacks, and hunt group calls.
- If the secondary extension button is unlit, pressing the button places an intercom call to the primary phone. If the button is flashing, an intercom call can still be placed without answering the incoming call by pressing the pound (#) button before the secondary extension button or by manually dialing the primary phone’s extension number.

There are two programmable options that determine the type of call indications the phone user receives for calls on a secondary extension button:

- **Ring When n Calls At Extension**: Allows the phone to receive a burst of ringing when a specified number (“n”) of calls queue-up at the associated primary phone. The ring signal repeats periodically as long as “n” number of calls are present at a primary phone. The Secondary Extension Key Alerting Tone timer determines the interval at which the ring signal repeats. If “n” is set to

0, the secondary phone never receives the burst of ringing. During keymap programming, the option is programmed individually for each secondary extension button.

- **Transient Call Indication On Call Answer**: Determines if the phone user sees a call display when answering a call by pressing the secondary extension button or reverse-transferring the call. If enabled, the display indicates if the call was ringing, recalling, transferred, or holding at the primary phone.

The secondary extension button can be located on the phone. There are two ways to make a secondary extension button:

- Create a secondary extension button in the keymap and determine which phone it will represent. This will be the same for all phones assigned to that keymap.
- Set the default value of a user-programmable button to “secondary extension key” in the keymap. This allows each phone user assigned to that keymap to determine the associated phone for his or her secondary extension button.

User-programmable buttons cannot be made into secondary extension buttons by the phone user; they can only be created in the database. Also, the phone user cannot change the button into any other ***type*** of button (such as a Speed Dial or feature button), even though it is assigned to a user-programmable button.

Any phone button can be designated as a secondary extension button, but buttons with lamps are recommended so that the secondary extension button can show call status. For flash rate definitions, refer to “Specifications” chapter in the *MiVoice Office 250 Installation Manual*.

The secondary extension button shows only calls that appear under the CALL buttons at the primary phone. Any calls on individual trunk buttons at the primary phone are considered private calls and will not appear on the secondary extension button.

Normally, the secondary extension button will not show intercom calls. However, if the primary phone does not have an **IC** button, intercom calls will appear under a CALL button and, therefore, will appear on the secondary extension button. If the primary phone is a single line phone, ***all*** calls will appear on the secondary extension button.

After answering a call on the secondary extension button, the phone user can transfer it back to the primary phone using one of the following methods:

- **Transfer to hold**: Press the secondary extension button, announce the call, if desired, and then hang up.
- **Transfer to ring**: Press **TRANSFER** and then the secondary extension button. You can announce the call, if desired, before hanging up to complete the transfer.

Because the calls that appear on the secondary extension button are active only at the primary phone, they will not follow call forwarding or system forwarding paths at the phone with the secondary extension button.

Phone LED Indications

Phone CALL buttons, trunk buttons, Speed Dial buttons, hunt group buttons, and some of the feature buttons contain light-emitting diodes (LED). LED flash rates indicate the status of the phones, trunks,

and features. For details about LED flash rates, refer to the “Specifications” chapter in the *MiVoice Office 250 Installation Manual*.

Because some features and phone buttons are not supported in SIP mode, the LED indicators and flash rates vary. For example, when a user places a call on hold in ITP IP mode, the Call or IC button LED uses the flutter flash to indicate that the call is on hold. In contrast, when the user places a call on hold in SIP mode, no LED indication appears because SIP does not support the CALL or IC buttons.

Red and Green LEDs

Mitel digital telephones have red and green LEDs in the button lamps. When both colors are used, a green lamp generally indicates that the call is active, ringing, or holding at your phone (see below for a possible exception for direct ring-in calls). When the lamp is red, the call is active, ringing, or holding at another phone. Trunk access, allowed-answer permission, and the feature being used (for example, System Hold or Call Transfer) determine if you can pick up a (red) call from another phone.

For more information about keymaps, refer to the “System Settings” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Two system-wide programming options control these LEDs.

- **Allow Green LEDs**: Gives the installer the option of having all red/green phones and red-only phones operate the same, or allowing the red/green phones to use the green LEDs. If this flag is enabled, the dual-color phones will use both the red and green LEDs. If it is disabled, the phones will operate in red-only mode, like the phones without green LEDs. In the default state, the flag is disabled.
- **Use Green LEDs For Direct Ring**: Determines whether direct ring-in calls cause a red or green lamp. If the flag is enabled, direct ring-in calls to your phone will be green. If disabled, all direct ring-in calls will cause the LED to be red. In the default state, this flag is enabled. The Allow Green LEDs flag must be enabled in order for this flag to operate.

Headsets

For convenience, an electret headset can be used at any phone. The Speaker button is used to connect and disconnect calls when the headset is attached. For more information about headset use, refer to the applicable phone user guide.

Headset Power Saver Mode

Some headsets have a power-saver mode that disables the headphone after a period of silence. To prevent these headsets from missing portions of incoming calls, there is a phone flag called “Send Alert Burst To Headset.” Enabling this flag allows the phone to generate a tone that activates the headset before connecting to an incoming call. In the default state, it is disabled. For more information, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Headset Connect Tone Phone Flag

The “Headset Connect Tone” flag sends a single tone instead of normal ring tones to a headset when a call is ringing at the phone with a non-handsfree intercom call, outside call, callback queue, or reminder message. The tone is sent only once, when the call is received, and does not repeat even if the call continues to ring unanswered. It is intended as a “wake up” tone for headsets that have a power-saving function that allows the device to “sleep” if idle for a set period. The flag is programmed on a phone-by-phone basis. By default the flag is disabled.

Volume Controls

On all phones, volume is controlled through volume control buttons on the phone. Refer to the applicable phone user guide for more information about volume controls and feature use.

Phantom Devices

Phantom device functionality has been enhanced in v2.2. The new phantoms are fully functional virtual devices on the system. Phantoms can function with Unified Communicator (UC) to perform advanced call routing tasks without the need for a real desk phone. They can also have a true status, such as idle, Do-Not-Disturb (DND), ringing, etc. so that they could be placed in hunt groups and actually ring.

The statuses that do not apply to phantom devices are as follows:

- Unplugged (phantoms have no hardware and will always be online)
- Busy/Permanently Busy (phantoms cannot answer a call)

The user must have UC or another OAI application (for example, Contact Center Suite or Attendant Console) to fully take advantage of the phantom feature. Because the phantoms do not have corresponding hardware devices, all programming that cannot be done in DB Programming must be done from OAI/UC.

An OAI application can have phantoms perform almost all operations as a phone. The exceptions are listed below. These exceptions must be enforced because phantoms do not have voice paths.

- Phantoms cannot answer calls from an OAI application.
- Phantoms cannot make calls from an OAI application.
- Phantoms cannot be in a conference.
- Phantoms cannot reverse-transfer calls.

Some OAI applications, such as Attendant Console, can send feature code requests to devices. The following feature codes are not available for phantoms:

- Answer
- Auto Trunk Answer
- Barge-In
- Change Language
- Conference
- Emergency Call
- Page
- Page Receive On/Off
- Ring Tone Selection
- Queue
- Record-A-Call
- Redial

- Group Listen On/Off
- Handsfree On/Off
- Headset Off
- Headset On
- Headset On/Off
- Hookflash [Recall]
- Music
- Reverse Transfer
- Seize Device
- Steal
- System Hold
- Transfer
- Transfer To Hold
-

Phantoms consume the same software resources as a regular phone or IP phone, with the exception of the voice paths. Therefore, they can have an impact on system performance.

Phantoms in DB Programming are defaulted to the not-in-DND status. When upgrading an old database, if there were phantom devices in the old database, their DND statuses will not be changed after the conversion.

The following scenarios describe phantom configuration uses:

- A phantom can be configured as a stand-alone device. An employee could have a cell phone to use when they are not in the office but still need to be on the office switch. The administrator could program a phantom for them and the employee (or administrator) could forward the phantom to the cell phone, using UC. That way, callers could still call an extension on the office switch and be able to get the employee on his or her cell phone. In the extreme case, all devices on the switch could be phantoms for users with cell or other phones that are not physically connected to the switch.
- A phantom can be configured as a monitoring device. For instance, a user may put a phantom device in a hunt group and program a phone button to map to that phantom extension. Therefore, a user can monitor when the phantom is ringing, etc., and even answer the call either via a reverse transfer or a secondary extension key.
- A phantom can be configured as a call-park extension. The user may want to set up one or two (or more) phantoms to be extensions at which a call can be parked.
- A phantom can be configured as a general mailbox because phantoms can have associated mailboxes. When a caller calls the phantom, if the phantom is set up to go to voice mail, the caller can leave a message in this “general” mailbox. However, this configuration is not much different from an unassociated mailbox, with the exception that a phantom mailbox can be accessed off-node (whereas an unassociated mailbox cannot). Note that unassociated mailboxes can be accessed off-node, if a caller calls the appropriate voice mail application and enters the unassociated mailbox extension.

For programming instructions, refer to the “Phones and Devices” and “Troubleshooting” chapters in the *MiVoice Office 250 Features and Programming Guide*.

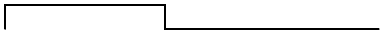
Signals and Tones

The MiVoice Office 250 provides auditory signals and tones that convey particular messages to users. This section describes the meanings of system sounds and graphically depicts what is heard by the user when the phone rings and what is heard over the handset during system operation.

Ringing Signals

The following table depicts the system signals and tones that are generated by the ringer or phone speaker, as shown in **Table 78**.

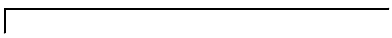
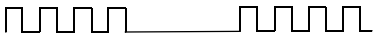
Table 78: Ringing Signals

Type of Call	Signal	
Outside call or recall	Single long tone every 6 sec.	
Intercom call, off-hook voice announce call, or queue callback	Two short tones and a pause (repeats for private calls and callbacks). ***NOTE:*** If the phone option for different IC/CO alerting tones is disabled, this will be the same as the outside call signal shown above.	
Reminder message	Eight short tones	

Call Processing Signals

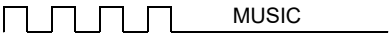
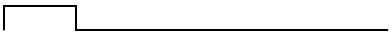
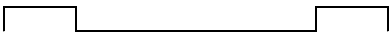
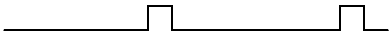
Table 79 shows the signals and tones that are heard through the handset or phone speaker.

Table 79: Call Processing Signals

	SIGNAL	ACTION
Intercom dial tone		Handset lifted or Speaker or IC button pressed.
Double tones (called phone receives the same signal)		Intercom call—repeats for private call or call to non-handsfree phone; one double tone for handsfree call.
Intercom busy signal — slow repeating tones		Called phone or selected trunk is busy.
A repeating signal of four fast tones and a pause		Called phone is in Do-Not-Disturb mode.
Recorder tones — fast tones (may be four or continuous)		Entry was not accepted, number is toll restricted, or dialing timer expired.

(Sheet 1 of 2)

Table 79: Call Processing Signals (Continued)

SIGNAL		ACTION
System busy signal — reorder tones followed by music		All resources are busy, phone is camped on.
Single tone		Confirmation that procedure was completed, or prompt that something needs to be entered.
Camp-on tone — single tone every 15 sec (programmable)		Another phone has camped on to this single line phone. (This tone is heard through the phone speaker only.)
Single line message waiting tone — six tones followed by intercom dial tone after lifting the handset or hookflashing		A message is waiting at this single line phone.
Single line call forwarding/Do-Not-Disturb tone — dial tone periodically interrupted by a single tone after lifting the handset or hookflashing		This single line phone has call forwarding and/or Do-Not-Disturb enabled.

(Sheet 2 of 2)

Phone-Supporting Features

The following sections describe the network and platform capabilities that support phone operation on the MiVoice Office 250.

Peer-to-Peer Audio

Peer-to-Peer (P2P) audio allows IP devices to transmit and receive audio directly with each other. The P2P feature reduces delay and packet loss for IP and SIP devices.

Devices that support P2P audio include:

- AudioCodes™ MP-114 or Quintum® AFT 400 SIP gateway and associated phones
- IP phones
- Multi-protocol phones

> **NOTE** 86xx multi-protocol phones in SIP mode are not supported on the MiVoice Office 250 platform.

To use P2P audio, phones must be programmed as members of the same Network Group. Calls between IP and SIP devices that are members of separate Network Groups cannot use P2P.

The Network Group Diagnostics feature allows an administrator to verify that the Network Groups are programmed correctly. For instructions on how to use the Network Group Diagnostics feature, refer to the *MiVoice Office 250 Endpoint and Voice Mail Administrator Guide* .

IP devices that use P2P audio do not use Base Server resources when they are connected in a P2P call. Therefore, the following features will *not* work during a P2P call:

- Agent Help
- Record-A-Call
- Station Monitor

NOTICE

Possible Performance Degradation. Passing real-time streaming data, such as audio, through encrypted virtual private networks (VPN) may significantly impact the network performance, router and firewall functionality, and audio quality.

IP Phone NAT Options

Automatic NAT Detection: The Automatic Network Address Translation (NAT) Detection feature allows Mitel IP phones to operate inside or outside a private network NAT or firewall without having to change the IP Settings\NAT Address Type field every time the phone is relocated. The feature enables Mitel hard IP phones and the Mitel 8602 IP SoftPhone to place and receive calls either from inside or outside a Mitel private network.

Network Address Translation (NAT) is a method for translating internal IP addresses or network addresses into public Internet addresses.

- **NAT Traversal for IP Phones**: The MiVoice Office 250 supports the connection of IP phones from the public network when the MiVoice Office 250 is located inside a local area network (LAN).

  To accomplish this near-end NAT traversal capability, the MiVoice Office 250 is programmed with both its Native IP address and the corresponding statically NATed public IP address. The platform being programmed with both addresses allows system-defined IP phones calling in from the Internet to traverse the NAT/firewall.

  You can program the MiVoice Office 250 so that each IP phone could use either the Native address or the NAT address. The Native NAT Type would be selected when the phone was located inside the NAT, and NAT would be selected when the phone was located outside the NAT. In other words, the MiVoice Office 250 could be programmed with corresponding IP addresses for communicating with each IP phone, whether the phone was located inside or outside the NAT/firewall.

- **Automatic Detection for NAT Traversal**: The Automatic NAT Detection feature takes near-end NAT traversal one step further. An IP phone designated with the Auto NAT Type automatically selects the appropriate NAT or Native IP address. This feature allows the IP phone to be moved from inside a NAT to outside and back again without programming intervention.

  If the default Auto NAT Type is designated for an IP phone, the Auto NAT Detection feature determines the correct IP address to use when the phone comes online. Once the NAT type is determined, the IP phone uses the setting for all calls. However, the NAT IP address associated with the system must be programmed manually in order for the NAT detection test to run.

  This feature does not work with IP Gateways, IP trunks, Mitel wireless 8664, 8665, and 8668 phones, any Mitel IP phones in SIP mode and third-party SIP phones, and Mitel phones. Firmware on these phones cannot run the Auto NAT Detection test. For these phones, the installer needs to manually select the NAT Type setting.

Data Port Modules

The following Data Port Modules can be connected to Executive, Professional, Standard, or Associate Display phones or to 8520 or 8560 phones:

- **PC Data Port Module (PCDPM)**: A digital device that is connected to the Executive Display, Professional Display, Standard Display, Associate Display, 8520, or 8560 Phones and supports data transmission between the phone and the phone system for DSS/BLF, off-hook voice announce, or Desktop Interface. A PCDPM is required if a Modem Data Port Module or DSS/BLF Unit will be installed, or if the user wishes to receive OHVA calls. A PCDPM is not required for the PKM 16.
- **Modem Data Port Module (MDPM)**: An analog device that can be used with a PCDPM and connects to a modem for data applications or to a single line phone. The port itself has an individual extension number and complete phone programming setup including its own trunk access and toll restriction information.

Special programming is required to enable the Executive, Professional, Standard, and Associate Display phones and 8520 or 8560 phones to support some of the PCDPM and MDPM functions. For more information, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

Digital Display Phone Desktop Interface

Desktop Interface is a phone-*level* communications protocol that allows a PC, attached to a PCDPM on an Executive, Professional, Standard, or Associate Display phone or on an 8520 or 8560 phone, to emulate a phone for call handling. This protocol allows devices such as PC phones, operator consoles, ACD supervisors, and integrated contact management tools to be developed for the MiVoice Office 250.

NOTE The Desktop Interface feature is required if attaching Desktop Interface devices to a digital telephone.

There can be as many Desktop Interface devices on the system as there are Executive, Professional, Standard, or Associate Display and 8520 and 8560 phone circuits. The only limitation is on Desktop Interface devices that require DSS/BLF status updates, such as operator consoles. The system supports up to 20 DSS/BLF status requests from external devices at one time. Therefore, if the device requires continuous DSS/BLF status updates, only 20 can be installed on the system at one time. Devices that require one-time or infrequent DSS/BLF updates are not limited by this capacity, except that only 20 requests for updates can be processed at one time. There is a limit of four virtual DSS devices per system because of the number of requests they require and the amount of information that is shared with the system.

Self-Programming Mode (8620/8622 and 8662 Phones Only)

Use the self-programming mode to configure the 8620/8622 and 8662 phones. This self-programming mode is similar to the 8660 IP phone. It allows you to set all the basic settings required for ITP mode.

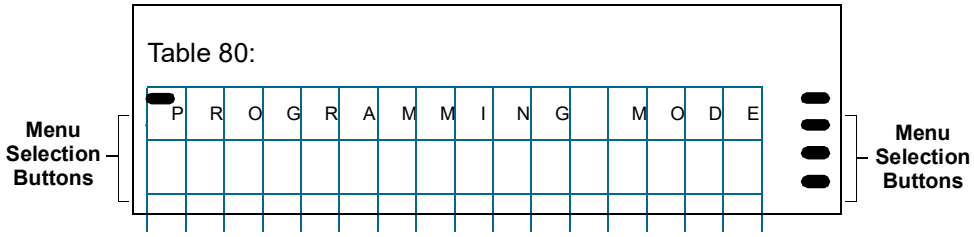
The following phones do ***not*** support self-programming mode:

- **8600 Phone**: This phone does not have a display. Hold **7** and **8** during power up to default its configuration.
- **8690 Phone**: Even though the 8690 phone has a virtual six-line display, it does not support self-programming mode because there are no physical buttons to hold during power up. Instead, you can configure the network settings through the Setup Wizard. For details, see page 552.

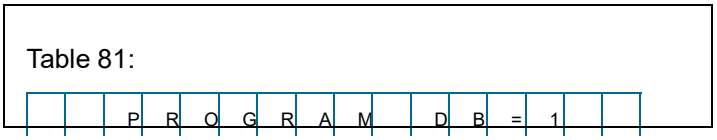
To use the self-programming mode on 8620/8622 and 8662 phones:

1. While pressing **7** and **8** together on your phone, unplug and replace the phone power cord. The phone enters the self-programming mode and the following screen appears on the phone display.

Six-Line Display for the 8662 Phone:



Two-Line Display for the 8620/8622 Phone:



2. Choose from the following options:
   - **Program Database**: Allows you to program the static network settings through a series of screens that follow (see page 532).
   - **Default Database**: Allows you to use predefined defaults for the settings (see page 533).
   - **Exit**: Allows you to exit the self-programming mode without making any changes (see page 533).

   With an 8662 phone, select one of the options using the menu selection button next to the LCD.

   With an 8620/8622 phone, press **1** to program the database or **2** to default the database. Or, you can use the lower six programmable buttons in both of the right-most. These buttons have the same functionality that the corresponding six-line display menu button would for each of the self-programming screens.

Program Database

Follow the displays and complete the programming. For details about **REVERT**, **EXIT**, **<<**, and **>>**, see page 533. Unlike the previous phones, such as the 8600 phone, multi-protocol phones do not support some configuration fields, such as hostnames and BOOTP. These phones, however, have additional fields like the software image information.

Table 82: Display Phone Self-Programming Prompts and Actions

When the phone screen displays . . .	Perform this action . . .
STATIC IP xxx.xxx.xxx.xxx	Enter the static IP address. Defaults to 192.168.200.201.
STATIC SUBNET xxx.xxx.xxx.xxx	Enter the default static subnet mask. Defaults to 255.255.255.0.
DEFAULT GATEWAY xxx.xxx.xxx.xxx	Enter the default static gateway address. Defaults to 192.168.200.201.
REMOTE IPRA IP xxx.xx.xx.xxx	*(ITP Mode Only)* Enter the remote IP resource application (IPRA) IP address. Defaults to 172.16.15.152.
REMOTE UDP PORT xxxx	*(ITP Mode Only)* Enter the remote UDP port. Defaults to 5567.
REMOTE TCP PORT xxxx	*(For ITP Mode Only)* Enter the remote TCP port. Defaults to 5566.
TFTP SERVER URL XXXXXXXXXXXXXXXX	Enter the URL of the TFTP Server (up to 79 digits). If the URL is too long to fit on the second line, it scrolls from right to left to allow the full URL to be displayed. ***To program the URL:*** The URL can contain any characters including the period (.) and hyphen (-) (press ***** and **#** respectively). Refer to the applicable phone user guide for instructions to enter characters. The slash (/) is used for the URL and the backslash (\) is used for the SIP authentication password.
PASSWORD * * * * * * * * * CHANGE CASE	*(For SIP Only)* Enter the authentication password—up to nine characters—for SIP registration. To enter the password, use the table shown above. To change the case of a character, select **CHANGE CASE**.
DHCP? xxxxxxxx	This option allows you to program the option to use DHCP. The second line from the top indicates the current status (either ENABLED or DISABLED). ***To program this option:*** Select either **ENABLED** or **DISABLED**. The phone updates the display to reflect the new state.
PHONE VLAN ID 0 DISABLE	Enter the VLAN ID of the phone port (0–4094). For more information, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

To program the IP address:

1. Press a dialpad button to enter the IP address. The entire second line, except the dots, disappears, and the number you entered appears in the first space.

   The following options are available:

   - **Revert**: Reverts to the old value found in the current database, and the phone updates the display accordingly.
   - **Exit**: Exits to the confirmation screen (last screen) without scrolling through all the other options. See **page 533** for more details on the confirmation screen.
   - **>>:** Continues to the next screen in the series.
   - **<<:** Returns to the previous screen in the series. Note that this option does ***not*** appear on the Static IP Programming screen.

2. After you have finished programming, select either **EXIT**, **>>**, or **<<**. The phone will check the field for errors.
   - *If there are no errors,* the phone displays the next programming screen.
   - *If there are errors,* the phone displays **INVALID ENTRY**. Correct the error by either selecting **REVERT** or modifying the incorrect entry.

Default Database and Exit

When you have scrolled through all the options, select **EXIT** from any of the screens or select **DEFAULT DATABASE** from the initial screen. The following screen appears to confirm the changes.

	S	A	V	E		C	H	A	N	G	E	S	?		
Y	E	S													
N	O														
<	<														

You have the following options:

- **Yes**: Overwrites the database. If you select this option, the **CHANGES SAVED** screen appears to notify you that the changes have been saved. To continue initializing the phone, press any button.
- **No**: Leaves the database as it was. If you select this option, the **NO CHANGES SAVED** screen appears to notify you that the changes have been discarded. To continue initializing the phone, press any button.
- **<<**: Returns to the previous screen in the series.

IP Phone Web Interface

The Web interface, IP Phone Web Client Administrative Session, for the multi-protocol phones allows you to set the most frequently used options. The Web interface also allows you to reset the device and synchronize its configuration with the configuration files on the TFTP server. Clicking **Synch** in the Configuration page synchronizes with the TFTP server only and ***not*** with the system. See **page 542**.

To reach the Web page on the 8690 phone, you must use a different computer than the 8690 phone. This is because the Windows CE .NET and the VPS (Voice Processing Subsystem) share an IP address and the Web pages are served from the VPS but browsed from the Windows CE .NET. After the Windows CE .NET sees its own IP address, the request never makes it to the VPS. When an external computer requests the Web pages, the VPS receives the request first so the Web pages are properly served. The Web Client configuration can be overwritten by a higher priority configuration source, such as self-programming mode.

NOTICE

The passwords, usernames, and URLs of the Administrative Web Sessions are all case-sensitive. Unless you enter the characters in the correct case (upper or lower), the browser will be unable to find the appropriate Web page.

To use the IP Phone Web Client:

1. Make sure the IP device is powered up, connected, and online.
2. Open Microsoft Internet Explorer and enter the phone IP address followed by **:8080** (for example, http://172.17.158.14:8080) in the Address field.
3. Log on with your username and password. The default username is **IPT** and password is **iptpassw**. To prevent unauthorized access, change the password of the IP Phone Web Client as soon as possible. **To change the logon password,** see page 539**.** The username is predefined and cannot be changed. The Web client contains the fields as described in **Table 83**. Some of the fields can also be programmed in a TFTP configuration file. For more information about TFTP configuration files, refer to the “Installation” chapter in the *MiVoice Office 250 Installation Manual*.

Table 83: IP Phone Web Client Field Descriptions

ITP Mode	Page #	SIP Mode
Device Configuration		
Network	**1-535**	Network
Audio & Call Control	**1-537**	Audio & Call Control
Phone	**1-538**	Phone
Trace	**1-539**	Trace
Authentication	**1-539**	Authentication
N/A	**1-539**	SIP
N/A	**1-540**	Feature Code
VLAN	**1-540**	VLAN

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Table 83: IP Phone Web Client Field Descriptions (Continued)

ITP Mode	Page #	SIP Mode
Status		
Network	**1-541**	Network
Firmware	**1-541**	Firmware
License	**1-542**	License
N/A	**1-542**	Volume
Image Upgrade	**1-542**	Image Upgrade
RTP/RTCP	**1-542**	RTP/RTCP
Command		
Configuration	**1-542**	Configuration
Reset	**1-542**	Reset

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Device Configuration

The Device Configuration field allows you to configure network, audio and call control, firmware, and trace settings.

Network: The Network Configuration page contains the network-specific configuration information. Both SIP and ITP modes have the same fields. The Network Configuration page contains the fields described in **Table 84**. To apply the changes, click **Update**.

Table 84: IP Phone Web Interface: Network Configuration Page

Field Name	Default	Indicates
MAC Address	xx-xx-xx-xx-xx-xx	The phone MAC address.
Static (Default) IP Address	192.168.200.201	The phone static IP address (when DHCP is disabled).
Default Network Subnet Mask	255.255.255.0	The phone default network subnet mask.
Default IP Gateway Address	192.168.200.201	The phone default IP gateway address.
IP Address Assignment	DHCP Enabled	Whether the IP phone uses DHCP to obtain an IP address instead of using the assigned static IP address.
TFTP Server IP Address	192.168.200.202	The IP address of the TFTP server where configuration files for this phone reside. Required if the DHCP does not provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is “192.168.200.150” and the path file is “/basic_sip,” then you can specify the URL as “192.168.200.150/basic_sip.”

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Table 84: IP Phone Web Interface: Network Configuration Page (Continued)

Field Name	Default	Indicates (Continued)
TFTP Update Interval	3600 seconds (one hour)	The time between synchronizations (in progress).
Enable TFTP Configuration Update	Enabled	Whether to perform the configuration updates from the TFTP server.
Use DHCP for TFTP Server and Path	Enabled	Whether to overwrite the TFTP URL in the TFTP Server IP Address field, listed above, with the TFTP server IP address and path from the DHCP server.

(Sheet 2 of 2)

Audio & Call Control: This Audio & Call Control Configuration page shows specific information about a call in progress. The information is only valid if there is a call in progress. Refreshing the page (by pressing the refresh icon on browser) while a call is in progress ensures that the information is current. Depending on the mode, the available fields are different.

ITP Mode: This page contains the fields described in **Table 85**. To apply the changes, click **Update**.

Table 85: IP Phone Web Interface: Audio and Call Control - ITP Mode

Field Name	Default	Indicates
Remote Server IP Address	0.0.0.0	The IP address of the IP resource application to which this IP phone is currently connected (the same value that is programmed in the system).
IP Terminal TCP Call Control Port	5566	The port number for TCP call control packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.
IP Terminal General Purpose UDP Port	5567	The port number for all UDP (audio) packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.

This page also shows the current call control status. The status fields are described in **Table 86**. To apply the changes, click **Update**.

Table 86: IP Phone Web Interface: Current Call Control Status Fields

Field Name	Default	Indicates
Remote Server IP Address	0.0.0.0	The IP address of the IP resource application to which this IP phone is currently connected (the same value that is programmed in system).
IP Terminal TCP Call Control Port	5566	The port number for TCP call control packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.

(Sheet 1 of 2)

Table 86: IP Phone Web Interface: Current Call Control Status Fields (Continued)

Field Name	Default	Indicates
IP Terminal General Purpose UDP Port	5567	The port number for all UDP (audio) packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.
RTP Send State	OFF	Whether the phone is currently sending RTP audio packets.
RTP Receive State	OFF	Whether the phone is currently receiving RTP audio packets.
Session Description	2025 (GW) 1000	A description that indicates the two phones on the current connection along with the connection type. Connection types include: • GW – gateway (using backplane) • PTP – peer-to-peer (not using backplane)
Audio Stream Transmit IP Address	xxx.xxx.xxx.xxx	The IP address of the IP device to which the phone is currently transmitting audio. This device may be another IP phone or an IP resource application.
Audio Stream Transmit Port	5006	The port number on the other IP device to which the IP phone is currently transmitting audio packets.
Audio Stream Receive Port	5006	The port number on this IP phone that is currently receiving audio packets.
Audio Frames/IP Packet	3	The number of audio frames that this phone is currently sending per IP packet. A higher value indicates a lower bandwidth, but an increased latency.
Speech Encoding Setting	G.729	The type of encoding that this phone uses when encoding DTMF tones.
Voice Activity Detection	Disabled	Whether the phone is currently transmitting RTP audio packets during periods of silence.

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SIP Mode: This page contains the fields described in the following table. To apply the changes, click **Update**.

Table 87: IP Phone Web Interface: Audio and Call Control - SIP Mode

Field Name	Default	Indicates
Number of Audio Frames	3	The number of audio frames per output audio RTP packet.
Codec Type	G.729 Annex A	The encoding/decoding codec preference for this phone.
Audio Port	5004	The port where the phone transmits and receives RTP packets. This port must be an even number; each connection uses the next available even-numbered port.
Diffserv	46	The IP precedence value.
Suppress Silence	Disabled	Specifies whether or not the phone sends audio packets during periods of silence.
Use Out-Of-Band DTMF	Enabled	Whether the phone is currently sending DTMF tones “out-of-band.”

Phone: The Phone Configuration page shows information related to the phone setup and whether the phone is currently in SIP mode. This page contains the fields described in **Table 88**. To apply the changes, click **Update**.

Table 88: IP Phone Web Interface Configuration Page

Field Name	Default	Indicates
Image Version	N/A	The image version of the phone-specific code (for example, *xxxx*_image_ver=1-2-13 (where the *xxxx* indicates the model name, such as the 8662 phone).
Image URL	xxx.xxx.xxx.xxx/image/image-<ss>.bin. The <ss> is the image version from "xxxx_image_ver."	The IP address or URL (up to 79 digits) of the image TFTP server. Required if the DHCP does not provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is "192.168.200.150" and the path file is "/image," then you can specify the URL as "192.168.200.150/image/image-1-3-4.bin."
IC Dial Plan	xxxx	*(For SIP Mode Only)* The dial plan mask that this phone uses to determine valid dial strings for end of dialing for IC, trunk, or outgoing calls. If the syntax is bad, you cannot dial calls. The acceptable symbols for a dial plan are described in the following table.
Trunk Dial Plan	9xxx	
Outgoing Dial Plan	[2–9]xxxxxxxx|1[2–9]xxxxxxxx|011[1–9]xxxxxxxxxxxxxx (This means that a dial plan can accept any 10-digit dial strings that begin with a "2" up to "9," any 11-digit dial strings that begin with a "1" and any digit "2" through "9," or any 18-digit dial strings that begin with "011" and any digit "1" through "9.")	
Hold Reminder Time	20	*(For SIP Mode Only)* The time from when a call was put on hold until the reminder or recall splash tone occurs.
SIP Mode	Disabled	Whether the phone will be in SIP mode or ITP mode after the next reset. The preferred method for changing between SIP and ITP mode is through the Web interface.

Symbol	Meaning
0–9	Digit 0–9.
x	Any digit 0–9.
*, #	Dialpad entry (*, #).
|	Expression inclusive OR.
+	Zero or more of the preceding digit(s) or [ ] expression. For example, "+#" means that a dial plan can accept any dial string that ends with "#," such as "1234*#" or "#" (zero digits). Also "[2-9]+#" means that a dial plan can accept any dial string that begins with any digit "2" through "9" followed by "#."
[ ]	Symbol inclusive OR.
-	Used only with [ ]. Represents a range of acceptable symbols.

Trace: The Trace Configuration page allows you to setup a remote trace capture for UDP debug information. The available fields vary depending on the mode the phone is in. This page contains the fields described in **Table 89**. To apply the changes, click **Update**.

Table 89: IP Web Interface—Trace Options

Field Name	Default	Indicates
Trace Host	0.0.0.0	The address of the remote device where trace output should be sent.
Trace Host Port	0	The port number of the remote device where trace output should be sent.
Trace SIP	Disabled	*(For SIP Mode Only)* Whether the trace output includes SIP information.
UDP Trace Log	Disabled	Whether the trace output includes UDP log information.
Telnet/Serial Trace Log	Disabled	Whether the trace output includes Telnet/serial log information.

Authentication: The Authentication page allows you to change the password to access the IP Phone Web Client. Passwords are case sensitive.

To change the password:

1. Enter the current password in the **Current Password** field, if there is one.
2. Enter the new password once in the **New Password** field and again in the **Confirm New Password** field.

To change the SIP authentication password:

1. Enter the authentication name.
2. Enter the current password in the **Current Authentication Password** field, if there is one.
3. Enter the new password once in the **New Authentication Password** field and again in the **Confirm New Authentication Password** field.

To apply the new password, click **Update**.

SIP: The SIP Call Client Configuration page shows SIP-related information. This page is only available when the phone is in SIP mode. This page contains the fields described in Table 90. To apply the changes, click **Update**.

Table 90: SIP Call Client Configuration Page Descriptions

Field Name	Default	Indicates
Phone Extension/Name	1000	The name used by the phone for authentication with the SIP proxy server.
Display Name	1000	The user or registration name on the phone that is shown if the phone has a display.
Proxy Server IP Address	192.168.200.100	The IP address of the primary SIP proxy Server.
Proxy Server Port	5060	The port of the primary SIP proxy Server.

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Table 90: SIP Call Client Configuration Page Descriptions (Continued)

Field Name	Default	Indicates
Register Server IP Address	192.168.200.100	The IP address of the primary SIP registration Server.
Register Server Port	5060	The port of the primary SIP registration Server.
Register Period	3600 seconds (one hour)	The time between synchronizations (in progress).
Voice Mail	VOICEMAIL	The voice mail string that is displayed when the **MSG** button is pressed or the Message feature code is entered.

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Feature Code: The Feature Code page shows the setting for the SIP mode feature codes. The available features and their default feature codes are shown in **Table 91**. For details about each feature, refer to the appropriate phone User Guide for SIP Mode. This page is only available when the phone is in SIP mode.

To apply the changes, click **Update.**

To default the feature codes, click **Default**.

Table 91: SIP Mode Feature Codes

Feature	Default Feature Code [Europe]	Feature	Default Feature Code
Answer	351	Hold	336
Attendant	0	LCD Contrast Adjustment	303
Conference	5	Message	365
Do-Not-Disturb On	370	Mute	314
Do-Not-Disturb On/Off	372	Outgoing	8
Do-Not-Disturb Off	371	Redial	380
Emergency	911 [999]	Redirect Call	331
Forward All	355	Reverse Transfer	4
Group Listen	312	Ring Tone Select	398
Headset On	315	Show IP*	300
Headset On/Off	317	Show Version*	9928
Headset Off	316	Transfer	345

VLAN: The VLAN Configuration page allows you to specify the VLAN IDs of the downlink (PC) ports and phone (IP Phone) port. It also allows you to enable LAN QoS if the phone is connected to a switch that supports 802.1P.

To specify the VLAN ID, enter the VLAN ID for each port. The available ports vary depending on the IP phone.

To enable the LAN QoS feature, check **Enable LAN QoS**. The values of the QoS are pre-defined and cannot be changed.

For more information about phone VLAN settings, refer to the “Phones and Devices” chapter in the *MiVoice Office 250 Features and Programming Guide*.

To apply the changes, click **Update**.

Status

Network: The Network Status page shows network information. Both SIP and ITP modes have the same fields. The page contains the fields shown in **Table 92**.

Table 92: Network Status Indications

Field Name	Indicates
Physical Address	Phone MAC address.
IP Address	Phone IP address.
Subnet Mask	Network subnet mask.
Default Gateway	Network gateway.
DHCP Server	DHCP server IP address the phone obtains.
DNS Server	DNS server IP address the phone uses.
Link Status	Link status of ports 0 and 1.
Negotiation	Negotiation status of ports 0 and 1.
Speed	Connection speed of ports 0 and 1.
Duplex	Connection mode of ports 0 and 1.

Firmware: The Firmware Information page shows current firmware information. Both SIP and ITP modes use the same page. This page contains the fields described in **Table 93**.

Table 93: Firmware Information Page Descriptions

Field Name	Indicates
Application Name	Phone application type, such as the 8600, 8660, 8662 phones.
Image Version	Version of the application.
Phonex Version	Version of the Broadcom Phonex (Phone Exchange) software.
Date/Time	Current date and time.
Boot Code Version	Version of boot image version number.
Hardware Type	Model type.
Tool Information	Versions of the tools (for example, gcc, GNU assembler, etc.).

License Page: Displays the end user license agreement.

Volume: The Volume Status page shows current firmware information. This page contains the fields described in **Table 94**. This page is only available when the phone is in SIP mode.

Table 94: Volume Status Page Descriptions

Field Name	Indicates
Ringer	Volume level for the ring tone. The range is 1–8.
Speakerphone Call	Volume level for the speakerphone call. The range is 1–8.
Speakerphone Tone	Volume level for the speakerphone tone. The range is 1–8.
Handset Call	Volume level for the handset call. The range is 1–8.
Handset Tone	Volume level for the handset tone. The range is 1–8.
Headset Call	Volume level for the headset call. The range is 1–8.
Headset Tone	Volume level for the headset tone. The range is 1–8.

Image Upgrade: The Image Upgrade page shows current image version information and download status. This page contains the fields described in **Table 95**. Both ITP and SIP modes has the same fields.

Table 95: Image Upgrade Page Descriptions

Field Name	Indicates
Image Version	Image version of the phone-specific code.
Configuration Image Version	Version of the configuration image.
Download Status	Status of the TFTP download.
Last Download Error Page	Additional error information concerning the status of the last TFTP application image update. For more information, refer to the*MiVoice Office 250 Installation Manual*.

RTP/RTCP Page: Displays the status of a Real-Time Transport Protocol/Real-Time Transport Control Protocol stack.

Command

Configuration: The Configuration page allows you to synchronize the configuration.

To synchronize the configuration, click **Synch**.

To default the configuration, click **Default**.

Reset: The Reset Hardware page allows you to reset the phone.

To reset the phone, click **Reset**.

Phone Configuration Files

This section describes all of the parameters that are available in the global and device-specific configuration files. You may add or edit any of the parameters in the configuration files. The files contain the following areas:

- Network Settings on **page 543**
- ITP Mode Settings on **page 546**
- Software Images on **page 547**
- Trace Settings on **page 548**

For details on how to download configuration files from a TFTP server, refer to the “Installation” chapter in the *MiVoice Office 250 Installation Manual*.

The configuration files overwrite any information programmed in the system when the phone is in ITP mode. Or, in the Setup Wizard on the 8690 phone, see page 552.

A phone in ITP mode receives configuration information from the system only after the phone comes online. The phone ignores any of the basic settings as if the “overwrite local settings” was permanently set to “disabled” for this circuit on the system.

The format of the configuration files is the Windows `.ini` format, “field name: value”. Although both the device-specific and global configuration files can contain any subset or all of the configuration fields, you only have to include the fields needed to configure the device.

For the phone to function in ITP mode, the following basic settings are required:

- The phone network settings (IP address, subnet mask, default gateway) including DHCP enable/disable.
- IP address and TCP port number of the IPR to which the phone belongs if not on the same LAN.
- Logon password (***not*** the same as the phone password).

For the phone to function in SIP mode, the following basic settings are required:

- The phone network settings (IP address, subnet mask, default gateway) including DHCP enable/disable
- Phone extension or registration name
- Authentication name and password

Network Settings

The configuration files contain the network parameters as described in **Table 96**. Some of the parameters can also be programmed in the IP Phone Web Client Administrative Session. The IP Phone Web Client field names and their reference pages that correspond to the parameters are also shown in **Table 96**.

Table 96: IP Phone Web Client Field Names—Network Settings

Parameter	Default	Indicates	IP Phone WEB Client (Page #)
dhcp	1	Whether to contact the DHCP server to obtain values for network-related parameters such as IP address, network gateway IP address, subnet mask, NTP, TFTP, domain name, DNS, VLAN ID, and more. The options are: • 0 – No DHCP server. Use static values. • 1 – Contact the DHCP server to obtain the values	See “IP Address Assignment” on page 535.
static_ip	192.168.200.201	The phone static IP address (when DHCP is disabled).	See “Static (Default) IP Address” on page 535.
static_snm	255.255.255.0	The phone default network subnet mask.	See “Default Network Subnet Mask” on page 535.
static_gw	192.168.200.201	The phone default IP gateway address.	See “Default IP Gateway Address” on page 535.
tftp_url	192.168.200.202	The IP address of the TFTP server where configuration files for this phone reside. Required if the DHCP does not provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is “192.168.200.150” and the path file is “/basic_sip,” then you can specify the URL as “192.168.200.150/basic_sip.”	See “TFTP Server IP Address” on page 535.
http_port	8080	The TCP port number of the Web server. A zero value disables the HTTP server.	N/A
opern_mode	1	• Whether the phone will be in SIP mode or ITP mode after the next reset. The preferred method for changing between SIP and ITP mode is through the Web interface.	See “SIP Mode” on page 538.
update_interval	3600 sec (1 hour)	The time, in seconds, between automatic configurations updates from the TFTP server. The interval should vary among phones to prevent massive simultaneous updates. The range of the values are 0-4294967295. A zero value disables the automatic update.	N/A
static_domain	0	The static domain name of the network if DHCP is ***not*** used.	N/A
static_dns_ip	0.0.0.0	The IP address of the DNS server if DHCP is ***not*** used.	N/A

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Table 96: IP Phone Web Client Field Names—Network Settings (Continued)

Parameter	Default	Indicates (Continued)	IP Phone WEB Client (Page #)
static_ntp_ip	0.0.0.0	The IP address of the NTP server if DHCP is ***not*** used. DHCP may also supply a NTP server. If this parameter has an IP address, it overwrites the DHCP NTP server IP address.	N/A
tftp_update_enable	1	Whether to perform the configuration updates from the TFTP server. The options are: • 0 – Disable • 1 – Enable **Note:** If this flag is disabled (set to 0) and the DHCP server provides the TFTP IP address and path, the phone will not start the update.	See "Enable TFTP Configuration Update" on page 536.
tftp_url_dhcp	1	Whether to overwrite the TFTP URL in the TFTP Server IP Address field, listed above, with the TFTP server IP address and path from the DHCP server. The options are: • 0 – Ignore the TFTP server IP address and path from the DHCP server. • 1 – Program the TFTP server IP address and path from the DHCP server to the tftp_url configuration parameter on the phone.	See "Use DHCP for TFTP Server and Path" on page 536.
lan_pq_port	0x0000	Specifies which of the Ethernet ports support 802.1P/Q. A "1" bit indicates that the port supports 802.1P/Q. **Note:** This parameter does not apply to phones with v1.1.x or later firmware. For the 8690 phone: • Bit 0x0001 – LAN/PWR Port • Bit 0x0002 – P1 Port • Bit 0x0004 – P2 Port • Bit 0x0008 – P3 Port For 8620/8622 and 8662 phones: • Bit 0x0001 – LAN/PWR Port • Bit 0x0002 – PC Port For the 8600 phone: • Bit 0x0001 – LAN/PWR Port	VLAN Configuration (see page 540)
telnet_port	23	The port number of the Telnet server. A zero value disables the Telnet server.	N/A

Table 96: IP Phone Web Client Field Names—Network Settings (Continued)

Parameter	Default	Indicates (Continued)	IP Phone WEB Client (Page #)
vlanid	0	Specifies the VLAN ID of the phone port. Do not use the configuration file to program the VLAN IDs for the 8690 phone. The configuration file update does not overwrite the Windows CE configuration, therefore, the changes will not be applied to the 8690 phone. To program the 8690 phone, use the 8690 Client Application. See page 552 for details.	VLAN Configuration (see page 540)
dwnlnk_vlanid_1	0	Specifies the VLAN ID of the PC downlink port.	
dwnlnk_vlanid_2	0	Specifies the VLAN ID of the PC downlink port 2 of the 8690 phone.	
dwnlnk_vlanid_3	0	Specifies the VLAN ID of the PC downlink port 3 of the 8690 phone.	
network_flags	0x00000000	Specifies whether or not the LAN QoS feature is enabled. The available options are: • 0x00: Disables LAN QoS. • 0x01): Enables LAN QoS.	VLAN Configuration (see page 540) .

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ITP Mode Settings

The configuration files contain the ITP mode-related parameters as described in**Table 97**. Some of the parameters can also be programmed in the IP Phone Web Client Administrative Session. The IP Phone Web Client field names and their reference pages that correspond to the parameters are also shown in **Table 97**.

Table 97: IP Phone Web Client Field Names—ITP Mode Settings

Parameter	Default	Indicates	IP Phone WEB Client (Page #)
cc_srv_ip	xxx.xxx.xxx.xxx	The IP address of the IP resource application to which this IP phone is currently connected (the same value that is programmed in the system).	See “Remote Server IP Address” on page 536.
cc_srv_tcp_port	5566	The port number for TCP call control packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.	See “IP Terminal TCP Call Control Port” on page 536.
cc_srv_udp_port	5567	The port number for all UDP (audio) packets that are transmitted from this phone for the current connection. This value may or may not be what is programmed in the system.	See “IP Terminal General Purpose UDP Port” on page 536.

Software Images

The configuration files contain the phone parameters as described in **Table 98**. Some of the parameters can also be programmed in the IP Phone Web Client Administrative Session. The IP Phone Web Client field names and their reference pages that correspond to the parameters are also shown in **Table 98**.

Table 98: IP Phone Web Client Field Names—Software Images

Parameter	Default	Indicates	IP Phone WEB Client (Page #)
xxxx_image_ver	xx-yy-zz	The image version of the phone-specific code (for example, xxxx_image_ver=1-2-13 (where the xxxx indicates the model name, such as the 8662 phone).	See "Image Version" on page 538.
xxxx_image_url	xxx.xxx.xxx.xxx/image/image-<ss>.bin. The <ss> is the image version from "xxxx_image_ver."	The IP address or URL (up to 79 digits) of the image TFTP server. Required if the DHCP does not provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is "192.168.200.150" and the path file is "/image," then you can specify the URL as "192.168.200.150/image/image-1-3-4.bin."	See "Image URL" on page 538.
8690_dsp_image_ver	N/A	(*For the 8690 Phone Only*) The call control firmware version string.	N/A
8690_dsp_image_url	N/A	(*For the 8690 Only*) The IP address or URL of the image TFTP server of the call control firmware image. Required if the DHCP does ***not*** provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is "192.168.200.150" and the path file is" /image," then you can specify the URL as "192.168.200.150/image/image-1-3-4.bin."	N/A
8690_os_image_ver	N/A	(*For the 8690 Phone Only*) The Windows CE firmware version string.	N/A
8690_os_image_url	N/A	(*For the 8690 Phone Only*) The IP address or URL of the image TFTP server of the Windows CE firmware image. Required if the DHCP does ***not*** provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is "192.168.200.150" and the path file is "/image," then you can specify the URL as "192.168.200.150/image/image-1-3-4.bin."	N/A

(Sheet 1 of 2)

Table 98: IP Phone Web Client Field Names—Software Images (Continued)

Parameter	Default	Indicates (Continued)	IP Phone WEB Client (Page #)
8690_gui_ image_ver	N/A	(*For the 8690 Phone Only*) The user interface firmware version string.	N/A
8690_gui_ image_url	N/A	(*For the 8690 Phone Only*) The IP address or URL of the image TFTP server of the user interface firmware image. Required if the DHCP does ***not*** provide the TFTP address. You can also include the path to the profile file, if desired. For example, if the TFTP server IP address is "192.168.200.150" and the path file is "/image," then you can specify the URL as "192.168.200.150/image/image-1-3-4.bin."	N/A

(Sheet 2 of 2)

Trace Settings

The configuration files contain the trace parameters as described in **Table 98**. Some of the parameters can also be programmed in the IP Phone Web Client Administrative Session. The IP Phone Web Client field names and their reference pages that correspond to the parameters are also shown in this table.

Table 99: IP Phone Web Client Field Names—Trace Settings

Parameter	Default	Indicates	IP Phone Web Client (Page #)
trace_ip	0.0.0.0	The address of the remote device where trace output should be sent.	See "Trace Host" on page 539.
trace_port	0	The port number of the remote device where trace output should be sent.	See "Trace Host Port" on page 539.
trace_flags	0x00000000	Enable specific trace features. • 0x01 – Turn on SIP traces. • 0x02 – Turn on UDP traces. • 0x10 – Turn on serial and telnet logs. *To turn on all of the trace features,* use the following value: 0x13.	 See "Trace SIP" on page 539. See "UDP Trace Log" on page 539. See "Telnet/Serial Trace Log" on page 539.

Export and Import Devices

When another node adds a new extension or changes an existing extension, username, or description, the network broadcasts the new extension information to the other nodes in the network. Or, system administrators can choose to export local device information to other nodes, or import devices from the other nodes to the local node, using the Export/Import option in this menu. If there is a conflicting (duplicate) extension in a broadcast from another node, the receiving node ignores the new extension in favor of its existing extension number.

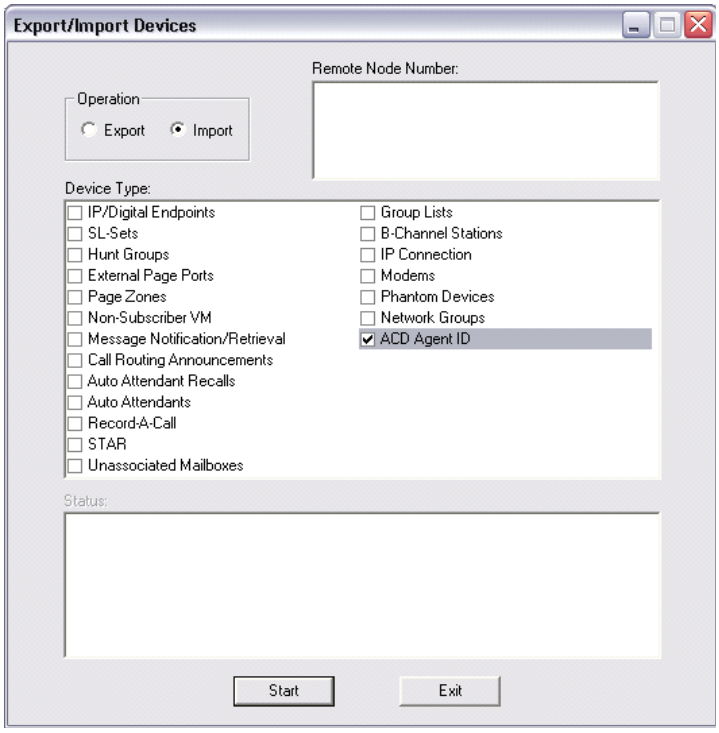
NOTE In the default state, the Voice Processor Enable flag is enabled. If you do not have an external Voice Processing system connected to the MiVoice Office 250, disable this flag before attempting to import or export information over the network.

If the network is unable to export or import an extension to a node (automatically or using the Export/Import feature) because there is an active programming session on that node, the node is unable to communicate with its Voice Processor port, the node is down, or the links to the node are down, the new extension will not be added to or changed on that node. (If Message Print is enabled, error messages will indicate any unsuccessful broadcasts.) You must manually add or change the new extension in the node database or try to export or import it later. You may want to check each node to verify that their off-node device lists are programmed properly to allow access between the nodes.

Extension numbers that exist before the network is established will not be automatically broadcast to other nodes, until they are modified. They must be exported or imported using the Export/Import option or programmed manually at each of the other nodes. Also, each node can have only 8000 off-node device entries in its database. When that limit has been reached, new devices received through network broadcasts cannot be added to that node database.

To export or import devices:

1. Start Session Manager. Then, from the DB Studio menu bar, select Operations and then **Export/Import Devices**. The following dialog box appears. You must have a programmed Node to export or import information from the node.



2. Select the following information, then click **Start**. Or, click **Exit** to cancel the operation without exporting any information.
   - **Operation:** Select the desired option to determine whether you will be importing devices into the local node or exporting local devices to a remote node.

- **Remote Node Number:** Select the node(s) you want to export the information to or import the information from, by placing checks in the appropriate boxes.
- **Device Type:** Select any combination of device types you want to export or import by checking the appropriate boxes. IP device information is automatically included if you import or export all phones. In addition, IP SLAs are automatically included if you import or export all single line sets.

NOTE ACD agents may be imported or exported the same as the other device types.

When you click **Start** to begin the export or import operation, the lower panel shows the export or import status.

If you are exporting information, the node you are exporting from is listed as “Node X: Export Source.” The destination node or nodes will show the node number and the current status of the export. Once the import from a node is completed, either successfully or unsuccessfully, the import source node displays the final status of the import. The screen will show messages as explained in **Table 100**.

Table 100: Node Status Descriptions

Status	Meaning
Starting Up	Indicates that the node is attempting to initiate the export.
Creation XX%	This node is XX percent done with the creation portion of the export.
Deletion XX%	This node is XX percent done with the deletion portion of the export.
COMPLETED	This node successfully completed the export.
Waiting...	When importing, this status indicates that this node is waiting for another node to finish before it can become an import source
Error – Not Reachable	The export to the node failed because the export source node could not communicate with this node.
Error – Phone Programming	The export to the node failed because someone on the node was performing Phone Programming.
Error – Voice Processor	The export to the node failed because the link to the Voice Processor is down.
Error – DB Programming	The export to the node failed because someone on the node was performing DB Programming.

3. After completing an import operation, terminate DB Programming to update the database.

If the local node cannot communicate with the remote node, the export or import will fail. Before you attempt to export or import information, verify that you can reach the remote node by dialing the extension assigned to the remote node. If you hear dial tone, you should be able to successfully export or import information. If you receive a NOT REACHABLE message, you must determine why calls are not properly being routed to the specified remote node before you can export or import information. If you camp on to a node while trying to reach the remote node, wait for the node trunk group to become available so that you can guarantee the remote node can be reached.

4. After attempting to export or import information, check Message Print output for error messages:
   - If a programming session is active on the remote node, you cannot export information to that node until the programming session has been terminated. Also, if the remote node has a Voice Processor and the link is down, the export will fail on the remote node.
   - If an existing extension on the remote node conflicts with an exported extension number, you will see one of the error messages listed in the following table. The error message indicates the resulting action on the remote node. In the example used in **Table 101**, the device information was exported from local node 1 to remote node 2.

Table 101: Example of Exporting an Extension from One Node to Another

Extension # Exported From Local Node 1	Existing Extension # ON Remote Node 2	Results on Remote Node 2
Phone 1000	Phone 10	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 100	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Phone 1000	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Off-Node Device ID(3:1000)	WRN DB CP Device Info Import For ID(1:'1000') Failed: Conflict with existing extension
Phone 1000	Unassociated Mailbox 1000	Off-Node Device ID(1:1000) created and Mailbox 1000 is associated with the off-node device
Hunt Group 2000	Unassociated Mailbox 2000	Off-Node Device ID(1:2000) created and Mailbox 2000 is associated with the off-node device
Phone 1000	Unassociated Mailbox 100	Off-Node Device ID(1:1000) created and Unassociated Mailbox is deleted
Phone 1000	Off-Node SL-Set ID (1:1000)	Off-Node Device ID(1:1000) deleted and Off-Node Device ID(1:1000) created
Page Port 91000 (99000 in Europe)	Unassociated Mailbox 91000 (or 99000)	Off-Node Device ID(1:91000 or 1:99000) created and Mailbox 91000 (or 99000) remains unassociated
Phone 1000	Off-Node Device ID (1:10XX)	WRN DB CP Device Info Import For ID(1:'1000') Failed: New extension matches existing wildcard

8690 Phone Programming

The following sections describe how to install the 8690 phone.

Windows CE .NET Programming (8690 Phone Only)

This section provides information about 8690 phone programming using the 8690 client application on Windows CE .NET. For complete information about how to use the 8690 client application, refer to the *8690 Administrator's Guide* (document part no. 550.8120) and the appropriate *Mitel Phone User Guide* (ITP or SIP mode). The following issues are discussed in this section:

- **Running the 8690 Setup Wizard** below.
- Upgrading the 8690 on **page 553**.
- Reformatting the 8690 Flash File System on **page 555**.
- Installing the 8690 on **page 556**.
- Configuring Phone Ports for a VLAN. Refer to the "Phones and Devices" chapter in the *MiVoice Office 250 Features and Programming Guide*.

Running the 8690 Setup Wizard

The first time the 8690 client application runs, the Setup Wizard launches. Once you successfully complete the Setup Wizard, it will ***not*** run again. The *Default Configuration* option on the Advanced menu resets the phone to its default state. If the Default Configuration option is run, the Setup Wizard will run automatically the next time the client application starts.

To run the Setup Wizard:

1. *EITHER*, Press **Network and Dial-up Connections** (under Windows CE .NET\Control Panel).

   OR, Press the Unified Communicator® (UC) Tray client icon ( ), then press Advanced – **Run Setup Wizard** from the menu. Press **Next** at the opening screen.

   NOTE UC is included in the 8690 client application package. This application provides simplified access to phone functionality, advanced phone features, and presence management options.

2. Select a network adapter from the list or accept the default adapter. Press **Properties**.
3. Press the option **Obtain an IP address via DHCP**. If the phone is not using DHCP, then specify the hostname or IP address of the phone, the subnet mask, the default gateway, and the DNS server. Then press **Next**.
4. In the Phone Configuration Download screen, enter the hostname or IP address of the computer hosting the phone configuration file in the format: 172.34.56.78.
5. At the UC screen, enter the following information. If you do not enter this information now, you will be prompted for it each time you log on or power on the phone.
   - **Web Address**: Enter the URL for the UC computer where your account is located.
   - **Username**: Enter your account username.
   - **Password**: Enter your account password.

- **Automatically log on**: Enable this option and you will not be prompted for logon information each time the phone starts or you select Log on from the Tray Client icon menu. Press **Next**.

6. Enable or disable the option to log on to UC in the Setup Wizard. Press **Finish**.

Upgrading the 8690

The 8690 client application can be upgraded automatically or manually. If an upgrade fails, or if the application is partially installed, you must reformat the flash file system on the phone. See page 555 for instructions on how to reformat the 8690 flash file system.

Upgrade the 8690 Automatically

The 8690 periodically checks for the following upgrades:

- **Firmware**: You are notified that the upgrade is starting, and the client application continues to run. You can monitor the progress of the upgrade on the six-line display. When the upgrade is complete, the phone resets.
- **Client Application**: The client application exits and the upgrade application launches. The upgrade application downloads the new version of the client application from the TFTP server, and you can monitor the download progress. When the download completes, the upgrade application launches the Windows CE .NET installation mechanism and installs the new client application version. At this point, you are prompted for where you would like to install the new files (the default is `\\Mounted Volume\8690 phone`).

  Press **OK** to continue with the installation. Press **Yes** when prompted to overwrite existing files. You are notified when the upgrade is complete. The upgrade application launches the new version of the client application and then exits.

  NOTICE

  Do not press **Cancel** when the installation dialog box prompts you to select a directory for the new files. At this point the previous client application has been deleted and if you cancel the upgrade, you will have to manually reinstall the client application. Also, do not change the default location (`\\Mounted Volume\8690 phone`) specified in the installation dialog box. The client application must be installed in the Mounted Volume folder to survive a reset or an upgrade to the operating system.

- **Operating System**: The client application exits, and the upgrade application launches. The upgrade application erases the existing version of the operating system from the persistent memory. Upon completion, the upgrade application downloads the new version of the operating system from the TFTP server. As the new version of the operating system is downloaded, it is simultaneously written to persistent memory, and you can monitor the progress. When the download and installation completes, you are notified and the phone resets. The registry settings are lost when the operating system is upgraded. This means that certain client application settings stored in the registry are lost as well. These include the session and diagnostic options.

When an upgrade is available, the Software Upgrade dialog box opens, and you are prompted to install the upgrade. To install the upgrade immediately, press **OK**. To install the upgrade later, you can set a reminder (1 hour, 1 day, or 3 days). When the reminder timer has expired, the Software Upgrade dialog box automatically prompts you to install the pending upgrade.

NOTICE

Possible Phone Inoperability. The 8690 does not have a built-in hard drive. Storage memory will ***not*** survive a reset or an upgrade to the operating system unless the files are stored in the Mounted Volume folder on the phone (My Computer\Mounted Volume) or on a host that is accessible from the 8690. Desktop shortcuts, and any preferences that have not been saved in the registry (including UC preferences) will be lost if you upgrade the operating system or if the phone resets. If you attempt to use the phone while an upgrade is in progress, the phone may be rendered unusable.

Upgrade the 8690 Manually

Although the 8690 automatically checks for upgrades, you can manually upgrade the 8690 by running the Auto Upgrade feature. If there are no new upgrades available, you will not be notified, and your phone will function normally. If there is an upgrade available, the Software Upgrade dialog box opens, and you are prompted to install the upgrade. You can choose to install the upgrade immediately by pressing **OK** or set a reminder to install it later.

To run the Auto Upgrade feature:

1. Press the System Tray icon, then press Advanced – Auto Upgrade – **Check for New Versions**. The upgrade search dialog box opens.
2. Press **OK** to allow the system to search for software upgrades. You are ***not*** informed if upgrades are *not* available.

In addition to running the Auto Upgrade feature, you can view, install, or cancel pending upgrades using the Advanced options.

To view and install pending upgrades:

1. Press the System Tray icon, then press Advanced – Auto Upgrade – **Pending Upgrades**. The Pending Upgrades dialog box opens and displays the status for the following:
   - Firmware
   - Client Application
   - Operating System

   The Upgrade Now option indicates a pending software upgrade.
2. Press **Upgrade Now** next to the upgrade you want to install. There is one button for each upgrade type.
3. When the Software Upgrade dialog box opens, press **OK** to install the upgrade.

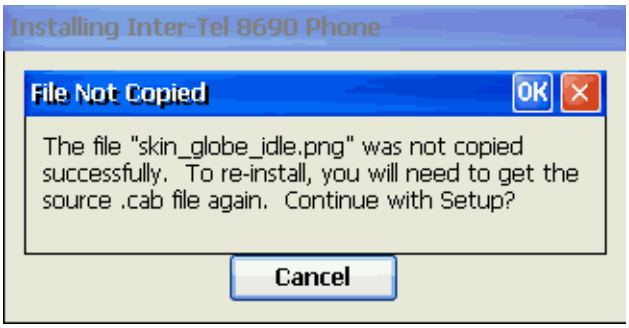
To cancel pending upgrades:

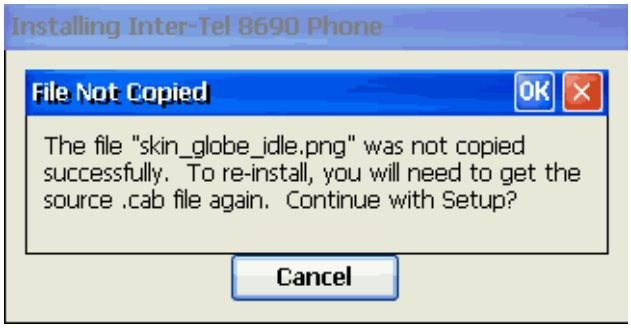
1. Press the System Tray icon, then press Advanced – Auto Upgrade – **Pending Upgrades**. The Pending Upgrades dialog box opens and displays the status for the following:
   - Firmware
   - Client Application
   - Operating System

2. Press **Cancel** next to the upgrade that you want to cancel. There is one button for each upgrade type.
3. Press **OK**.

Reformatting the 8690 Flash File System

Installing the client application on the 8690 may fail if the phone does not have sufficient memory resources or if the phone loses power before the upgrade is complete. If the upgrade fails, or if the application is partially installed (see the dialog boxes below), it is strongly recommended that you reformat the flash file system on the phone.



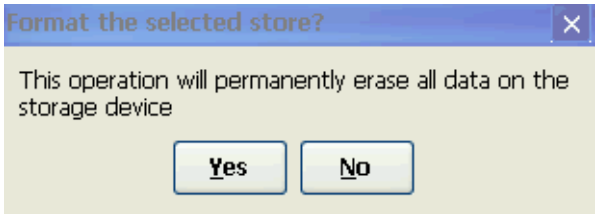


NOTICE

Data Loss Hazard. Any data or files saved in the Mounted Volume folder will be lost. To avoid loss of data, the data should be saved to a host, a memory card, or a location on the network.

To reformat the flash file system:

1. From the task bar, click Start – Settings – **Control Panel**.
2. Double-click **Storage Manager**.
3. Click **Dismount**, then click **Format**. The following dialog box appears.



4. Click **Yes** to erase all data.
5. Click **OK** to close the Format store succeeded dialog box.
6. In the Storage Properties dialog box, click **New** to create a new partition.
7. Enter a name for the partition, and then click **OK**.
8. Click **Properties**. The Partition Properties dialog box opens.
9. Click **Dismount**, and then press **Format**. The Format dialog box opens.
10. Verify that the Quick Format option is selected and click **Start**.
11. Click **Yes** to format.
12. Click **OK** to close the Format dialog box.
13. In the Partition Properties dialog box click **Mount** and then **OK**.
14. Close the Storage Properties dialog box and then exit the Control Panel.
15. Reinstall the client application as described on the following page.

Installing the 8690

The following lists the system requirements for installing the 8690 client application:

- Windows CE .NET v4.2. For details about Windows CE .NET Operating System, refer to the:
  - *MiVoice Office 250 Phone Administrator Guide*
  - *MiVoice Office 250 Unified Voice Messaging Administrator Guide*
- 400 MHz Processor, 128 MB RAM
- 5 MB Persistent Storage

The 8690 client application is distributed in a single CAB (cabinet) file. A CAB file contains one or more application files compressed into a single archive file (such as a WinZip® file).

To install the client application on the 8690:

1. Obtain the latest CAB file, **`8690Phone.8690_CEDOTNET_MIPSII.CAB`**, from the Mitel eDocs Web site (http://edocs.mitel.com).
2. Copy the CAB file onto your 8690 phone.
3. Double-click the CAB file to launch the installation program.
4. The display shows the default location, **`\Mounted Volume\8690 Phone`**, where the application will be installed. Click **OK** to accept the default location.

NOTE If you choose to install the product in some location other than the default, you must make sure that you install it in the flash file system (somewhere in the Mounted Volume folder). This ensures that the files will persist in the event that the phone is reset. When the installation completes, launch **`8690Phone.exe`** to start the client application.

If an installation or update fails, or if the application is partially installed, you must reformat the flash file system on the phone. See page 555 for instructions on how to reformat the 8690 flash file system.

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