MARCH NETWORKS™

3200

Integrated Communications Platform

General Information Guide

Release 2.3

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March Networks 3200 Wireless Applications Gateway
General Information Guide
PN 50001591, Rev. A
August 2001

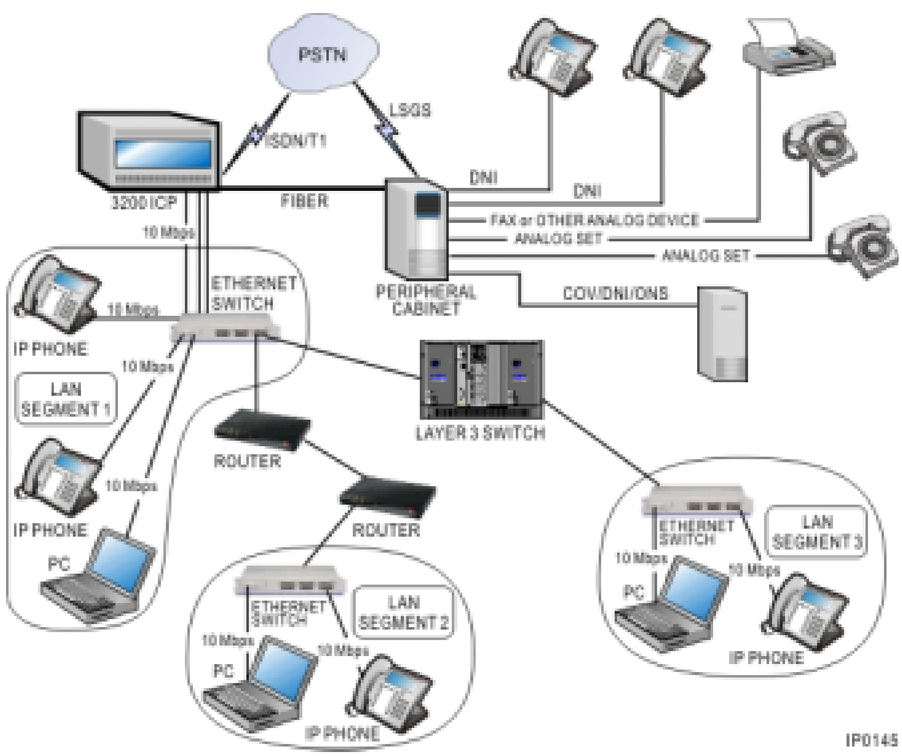
Table of Contents

Product Overview

The March Networks™ 3200 ICP system is the first integrated, single-server telecom solution to deliver full-featured telecommunications functionality in a Windows NT environment. Based on Mitel's proven PBX technology, the March Networks 3200 ICP system offers IT managers advanced telecommunications in a familiar, easy-to-manage LAN environment. Designed for companies with between 40 and 240 users, it supports both standard and enhanced PBX features and interfaces, including advanced call control, digital trunking, peripheral interfaces, auto-attendant, call center routing and management, computer telephony integration (CTI), wireless applications, a mobility solution, GUI-based management, and intelligent messaging functionality -- all pre-packaged and delivered on one server.

Far more than a voice application developed for NT, the March Networks 3200 ICP system is a feature-rich telecommunications application fully integrated into the computing environment.

The March Networks 3200 ICP system is a flexible, modular solution that allows organizations to add sophisticated functionality as their needs grow and change. For example, adding ISDN PRI connectivity or support for additional digital trunks is as easy as adding another card and updating software on the telephony server. Enhanced call center capabilities, directory services integration, and a host of other features are handled through simple software additions.



March Networks 3200 ICP System Architecture

LINES AND TRUNKS

LINES

The system supports the following types of internal voice lines:

- **Digital Network Interface (DNI) Lines** provide an interface for digital telephones, consoles, and datasets. The maximum loop resistance on a DNI line must be 280 ohms or less, and the loop length must be 3300 ft. (1000 m) or less on 26-gauge wire.

  These lines are supported by the Digital Network Interface line card.
- **On-Premises (ONS) Lines** (24 V per port) are for industry-standard rotary dial and DTMF telephones. The external loop resistance on an ONS line must be 600 ohms or less, and the loop length must be 5000 ft. (1500 m) or less on 26-gauge wire.

  These lines are supported by the ONS line card and the ONS CLASS/CLIP line card.
- **Off-Premises (OPS) Lines** (48 V per port) are for industry-standard telephones where the external loop resistance exceeds 600 ohms or where lightning surge protection is required. The maximum loop resistance on an OPS line must be 1800 ohms or less, and the loop length must be 19,000 ft. (5800 m) or less on 26-gauge wire.

  These lines are supported by the OPS line card.
- **Control Over Voice (COV) Lines** provide an interface for voice mail systems. The maximum loop resistance on a COV line must be 280 ohms or less, and the loop length must be 3300 ft. (1000 meters) or less on 26-gauge wire.

  These lines are supported by the COV line card.

TRUNKS

The system can connect to the public switched network or to private networks over both digital and analog trunks.

The following digital links are supported:

- **DS1 Links:** The system supports CO, DID, E&M, MSDN/DPNSS, and MSAN/APNSS protocols.

  The March Networks 3200 ICP system connects to DS1 links by using the Dual T1 card.
- **CEPT Links:** The system supports MSDN/DPNSS, and DASS II (UK only) protocols.

  The March Networks 3200 ICP system connects to CEPT links by using the Dual E1 card.
- **PRI Links:** The system supports DMS-250, DMS-100, Bellcore National ISDN, 4ESS, NI-2, 5ESS NI-2, and Euro-ISDN (CTR4) protocols.

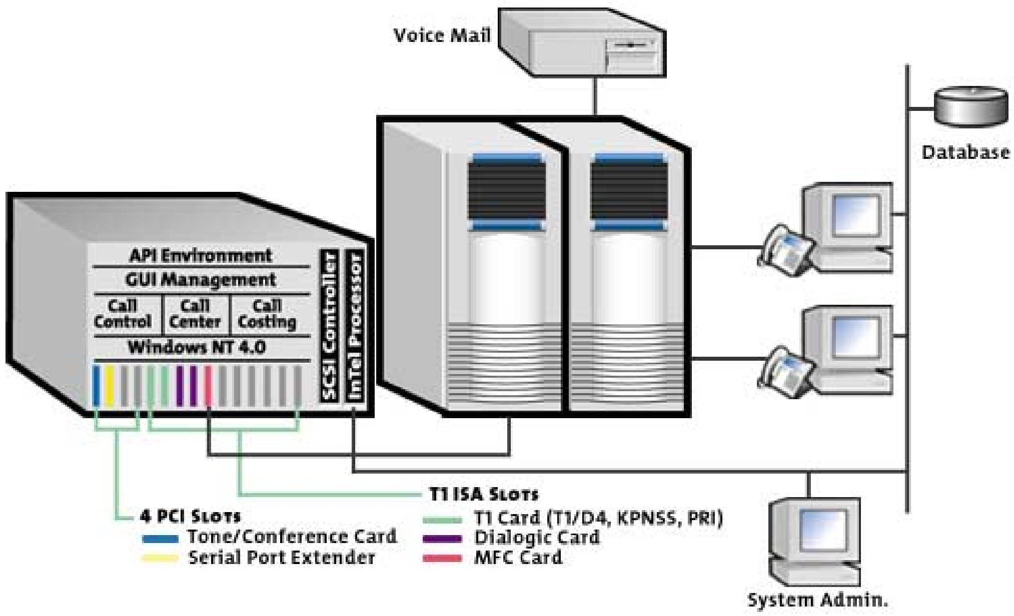
  The March Networks 3200 ICP system connects by using the Dual T1 or Dual E1 card.

The following analog trunks are supported:

- **Analog CO Trunks** interface to the system through the Loop Start/Ground Start (LS/GS) trunk card.

- **E&M Trunks** interface to the system through the E&M trunk card (which can be configured for either 2-wire or 4-wire operation). This card supports Type I through Type V circuits.
- **Direct Inward Dial and Tie Trunks** interface to the system through the DID/Loop Tie trunk card or the AC15 trunk card.

Supporting Applications



March Networks 3200 ICP Supporting Applications

Applications Support

The March Networks 3200 ICP system is capable of running the following application interfaces:

- MiTAI™ - The Mitel Telephony Applications Interface is a proprietary Applications Protocol Interface (API) that allows third-party-developed CTI applications to interface with the Mitel's call control. A programmer's toolkit plus run-time software is also available, which enables developers to create computer telephony applications.
- TAPI™ - Microsoft's TAPI is supported for desktop applications or client/server applications.
- For reliability purposes, all third party applications must run on an off-board server unless approved by Mitel Networks.

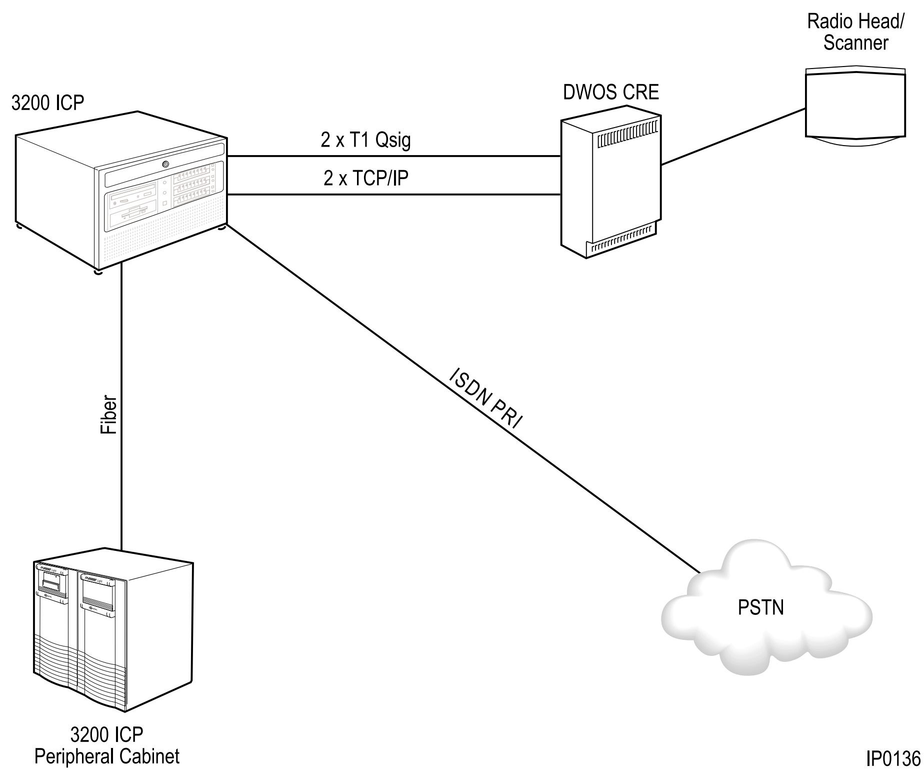
3800 Ericsson Mobile Advantage Gateway system

Ericsson Inc. and Mitel Networks have partnered to deliver a new integrated digital wireless office solution with PBX functionality, the 3800 Ericsson Mobile Advantage Gateway system. This integrated system combines Ericsson's Mobile Advantage™ Wireless Office (DWOS) and the 3200 ICP system to address the needs of both wired and wireless voice communication services in small to medium environments. The

3800 Ericsson Mobile Advantage Gateway system provides users with a mobile phone and one number for all their communication needs, regardless of location. Intelligent network features allow PBX numbering plans to be used even while roaming, thereby creating a wireless office within the wired environment.

The 3800 Ericsson Mobile Advantage Gateway system is a fully digital mobile communications system that acts as an extension of the wired 3200 ICP system. Inside the office using a mobile phone, the system creates a private cellular network. Outside the office, the system works via the public cellular network. All calls to a user's office number are routed automatically to the mobile phone whether inside or outside the office. Cellular rates only apply when a call is made via the public cellular network. External phone calls that are made on the mobile phone within the local PBX area are charged a flat rate determined by the carrier. The 3800 Ericsson Mobile Advantage Gateway system increases productivity by making employees significantly more accessible. The system operates on either 800MHz or 1900MHz licensed frequencies that are allocated according to availability.

The March Networks 3800 Ericsson Mobile Advantage Gateway system allows the wireless Ericsson™ phones to implement standard and enhanced system features. The 3800 Ericsson Mobile Advantage Gateway system can be clustered using the standard Mitel DPNSS Networking to communicate between multiple cellular phones working on different network nodes.



Physical Components of Ericsson Mobile Advantage™

The 3800 Ericsson Mobile Advantage Gateway system is comprised of the March Networks 3200 ICP system server and several Ericsson hardware components as described below.

March Networks 3200 ICP system server

The DWOS Mobility Server is the control center for the system. It connects to the Cellular Radio Exchange (CRE) for configuration and administration of the Radio Infrastructure (RI) and real-time tasks such as call switching. Additionally, this server provides the interfaces to the SS7 Gateway and a corporate LAN.

Central Radio Exchange

The Cellular Radio Exchange (CRE) allows connections between mobile phones operating in the system coverage area to be switched within the Mobile Advantage system without accessing external networks. The Cellular Radio Exchange is directed by the 3200 ICP system and provides an interface between the mobile phones and the wired system which connects to the external PSTN.

Scanners

Scanners are physically connected to the Cellular Radio Exchange and provide frequency information for neighboring Public Land Mobile Network (PLMN) cells to the Mobile Advantage system. The information obtained by Scanners allows the DWOS to minimize interference from the Mobile Advantage system on PLMN frequencies by the selection of unused allocated frequencies.

Radio Heads

Radio Heads (RH) connect by a Radio Link (RLINK) to the Cellular Radio Exchange for communications with Mobile Advantage mobile phones. Radio Heads are placed at positions within the Mobile Advantage system area to provide maximum coverage and capacity. The addition of multiple redundant Radio Heads in an area will increase call handling capabilities in addition to assuring continued call processing capabilities in the event of a unit Radio Head failure.

Mobile Phones

Mobile Advantage mobile phones are cellular or PCS phones that are compatible with TDMA ANSI-136 standard protocol and capable of ACELP speech encoding. These phones must be enabled for Public Service Profile (PSP) and Private Operating Frequency (POF). Mobile phones are external to the 3800 Ericsson Mobile Advantage Gateway system.

Further information on the Ericsson Mobility Advantage System can be found in the Ericsson Mobile Advantage Wireless Office System Installation & Configuration Manual.

DNIC Set Restrictions in the 3800 Ericsson Mobile Advantage Gateway system

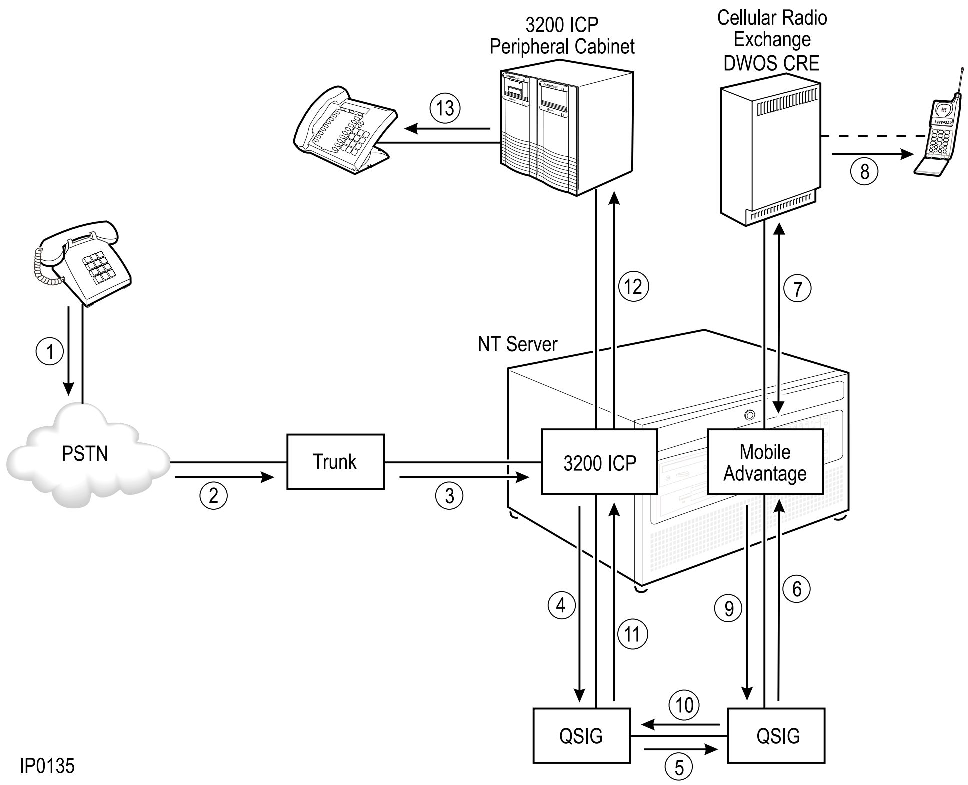
When cell phones are used together with DNIC phones in the 3800 Ericsson Mobile Advantage Gateway system, there is reduced functionality in the DNIC desktop set. This is a static and permanent condition within the system.

BASIC CALL CONFIGURATION FOR THE 3800 ERICSSON MOBILE ADVANTAGE GATEWAY SYSTEM

There are a number of scenarios that can occur within the 3800 Ericsson Mobile Advantage Gateway system context. For example:

- A call through the PSTN to 3800 Ericsson Mobile Advantage Gateway system (DNIC phone to DNIC phone with concurrent ringing of cell phone).
- A cell call through the Public Land Mobile Network (PLMN) to 3800 Ericsson Mobile Advantage Gateway system (Cell phone to DNIC phone with concurrent ringing of cell phone).
- An internal cell call from the 3800 Ericsson Mobile Advantage Gateway system to a cell phone in the PLMN.

The illustration below shows the pathway of a call from the PSTN to the 3800 Ericsson Mobile Advantage Gateway system with concurrent ringing enabled (desktop and Mobile Advantage cell phone):



The call pathway is explained in this manner:

1. A call is placed to the Public Switched Telephone Network (PSTN).
2. The PSTN takes the call and sends the call to a Trunk.
3. The Trunk routes the call to the NT Server where March Networks 3200 ICP system receives the call.
4. The call is then sent to March Networks 3200 ICP's QSIG.
5. The call is sent from March Networks 3200 ICP's QSIG to Mobile Advantage QSIG.

6. The call is then sent to Mobile Advantage.
7. The Mobile Advantage routes the call via T1 ISDN to the CRE. At the same time the Mobile Advantage also routes the call back through QSIG to the March Networks 3200 ICP system.
8. The CRE rings the TDMA cell phone and connection is established.
9. The call is routed back to the Mobile Advantage QSIG.
10. The call is sent from the Mobile Advantage's QSIG to March Networks 3200 ICP's QSIG.
11. The call is then sent to March Networks 3200 ICP system.
12. The call is sent to the Peripheral Cabinet.
13. The Peripheral Cabinet rings the desk phone and connection is established.

More information on the features and functions of the Ericsson Mobile Advantage Wireless Office can be found in the Mobile Advantage System Overview Manual.

Ericsson Wireless Assistant

Ericsson Wireless Assistant allows today's mobile users to be accessible wherever they are located using only one number. Being in contact at all times reduces the amount of time checking voice mail and returning phone calls which in turn increases the efficiency and effectiveness of employees. The ability for users to be reached by utilizing one number - a Personal Number- regardless of location is a key differentiator in today's fast moving mobile business environment.

In-Office Convenience

If the user is in the office, incoming calls can be routed to the desk phone and mobile phone simultaneously (Concurrent Ringing), allowing the user the convenience of answering wherever they are. Unanswered calls are routed to the location specified by the user, such as voice mail or another answering position.

Convenience and control are the key to the system. For example, there may be times when users do not want to receive incoming calls or only want to receive calls from certain key people. To achieve this, advanced call routing and control functionality is available.

Every user has access to a Personal Assistant, which can be accessed via a web browser or by dialing in from any touch-tone phone. The Personal Assistant allows users to define and/or change how their incoming calls are routed. Access to the Personal Assistant allows users to set up different routing plans for internal calls, external calls, and for calls from other numbers that the user can specify.

Out of Office Convenience

Traveling away from the office does not require any actions by the end-user to maintain accessibility while within the network coverage area. All calls are automatically routed to the handset, with minimum call-setup/ring time for the calling party. Users can also use Personal Assistant to control how their calls are routed. Additionally, for an unanswered call, the digital Personal Assistant can be set up to offer the caller a choice between being sent to voice mail or another phone in the office.

Personal Assistant

Although each user has several phone numbers (home phone, office phone, mobile phone, etc), callers only need to dial one number - the Personal Number - to reach the user. The Personal Assistant allows the user to create and manage up to four different Personal Number Profiles.These profile allows users to control how their calls are routed. The Personal Assistant can be accessed via a web interface or a touch-tone phone system. For more information, refer to the Wireless Assistant System Administration Manual and the Mobile Advantage Wireless Assistant End User Manual.

Configurations

There are two configurations of the Wireless Assistant functionality, the 3200 Integrated Communications Platform with Ericsson Wireless Assistant, page 10, and the 3800 Ericsson Wireless Assistant Gateway system, page 12.

3200 INTEGRATED COMMUNICATIONS PLATFORM WITH ERICSSON WIRELESS ASSISTANT

Using the Ericsson Wireless Assistant option on the 3200 ICP system requires the installation of Ericsson's Cellular Radio Exchange (CRE) with Switchboard.

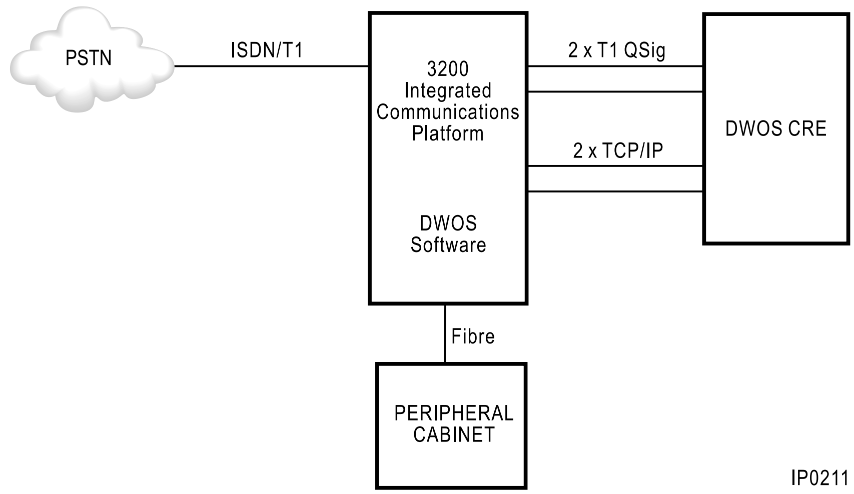
The 3200 Integrated Communications Platform is a Windows NT server-based switch controller. By enabling the Ericsson Wireless Assistant option, Ericsson's DWOS Mobility software offers enhanced mobility services for mobile business users via the intelligent integration of mobile phones and PBX extensions into one mobile solution. The 3200 ICP system with Ericsson Wireless Assistant connects to the switchboard for operations and maintenance of the system and switching of calls to, from and within the system. Network, system and end-user management are handled via a web-browser interface. The system also provides access to the enterprise voice mail system.

Cellular Radio Exchange with Switch Board

A sub-equipped Cellular Radio Exchange (CRE) with a power card and a switchboard connects the wireless system to the 3200 ICP system and directs call traffic. It contains a non-blocking switch, signal processor for tone detection, functions for voice announcements and primary rate interfaces (T-1 or E-1) on a single board. The switchboard has its own CPU and memory and acts as a self-sufficient processing node, networked with the 3200 ICP system via 10-BaseT Ethernet.

Radio Frequency Distribution (i.e. Macro Wireless Coverage)

This product uses existing Macro Wireless coverage to provide service to the wireless handsets. There is no additional Radio Frequency infrastructure included.



3200 ICP SYSTEM WITH ERICSSON WIRELESS ASSISTANT CONFIGURATION

DNIC Set Restrictions in the 3800 Ericsson Wireless Assistant Gateway system

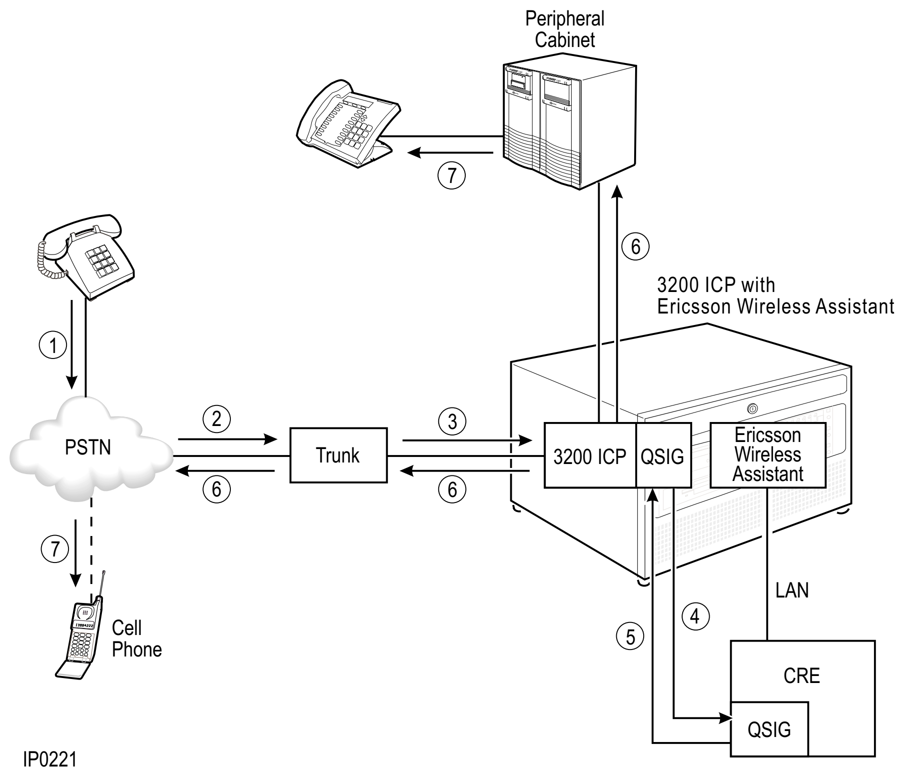
When cell phones are used together with DNIC phones in the 3800 Ericsson Wireless Assistant Gateway system, there is reduced functionality in the DNIC desktop set. This is a static and permanent condition within the system.

Basic Call Configuration for the 3200 ICP system with Ericsson Wireless Assistant

There are a number of call scenarios that can occur with the system. For example:

- A call through the PSTN to the 3200 ICP system (DNIC phone to DNIC phone with concurrent ringing of cell phone).
- A cell call through the Public Land Mobile Network (PLMN) to 3200 ICP system (Cell phone to DNIC phone with concurrent ringing of cell phone).
- An internal cell call from the 3200 ICP system to a cell phone in the PLMN.

The illustration below shows the pathway of a call from the PSTN to the 3200 ICP system with concurrent ringing enabled (desktop and cell phone):



The call pathway is explained in this manner:

1. A call is placed to the Public Switched Telephone Network (PSTN).
2. The PSTN takes the call and sends the call to a Trunk.
3. The Trunk routes the call to the 3200 ICP system.
4. The call is sent from 3200 ICP system QSIG to Wireless Assistant QSIG on the CRE.
5. The Wireless Assistant CRE then initiates two calls back over the QSIG to the 3200 ICP system (one for the desktop phone and one for the cell phone).
6. The call is sent to the Peripheral Cabinet and to the PSTN.
7. The Peripheral Cabinet rings the desk phone and the PSTN rings the cell phone.

More information on the features and functions of the Ericsson Wireless Assistant can be found in the Wireless Assistant System Installation & Configuration Manual and the Wireless Assistant System Administration Manual.

3800 Ericsson Wireless Assistant Gateway system

The 3800 Ericsson Wireless Assistant Gateway system offers enhanced mobility services for mobile business users via the intelligent integration of mobile phones and PBX extensions into one mobile solution. This configuration consists of two major hardware components, the March Networks 3800 Ericsson Wireless Assistant Gateway system and Ericsson's Cellular Radio Exchange (CRE) with Switchboard.

The 3800 Ericsson Wireless Assistant Gateway system is a Windows NT server-based switch controller, running March Networks 3200 Call Control Software and Ericsson's

DWOS Mobility software. The 3800 Ericsson Wireless Assistant Gateway system connects to the CRE switchboard for operations and maintenance of the system and switching of calls to, from and within the system. Network, system and end-user management are done via a web-browser interface. The 3800 Ericsson Wireless Assistant Gateway system also provides access to the enterprise voice mail system.

Cellular Radio Exchange with Switch Board

A sub equipped Cellular Radio Exchange (CRE) with a power card and a switchboard connects the system to the Ericsson Wireless Assistant Gateway system and directs call traffic. It contains a non-blocking switch, signal processor for tone detection, functions for voice announcements and primary rate interfaces (T-1 or E-1) on a single board. The switchboard has its own CPU and memory and acts as a self-sufficient processing node, networked with the 3800 Ericsson Wireless Assistant Gateway system via 10-Base T Ethernet.

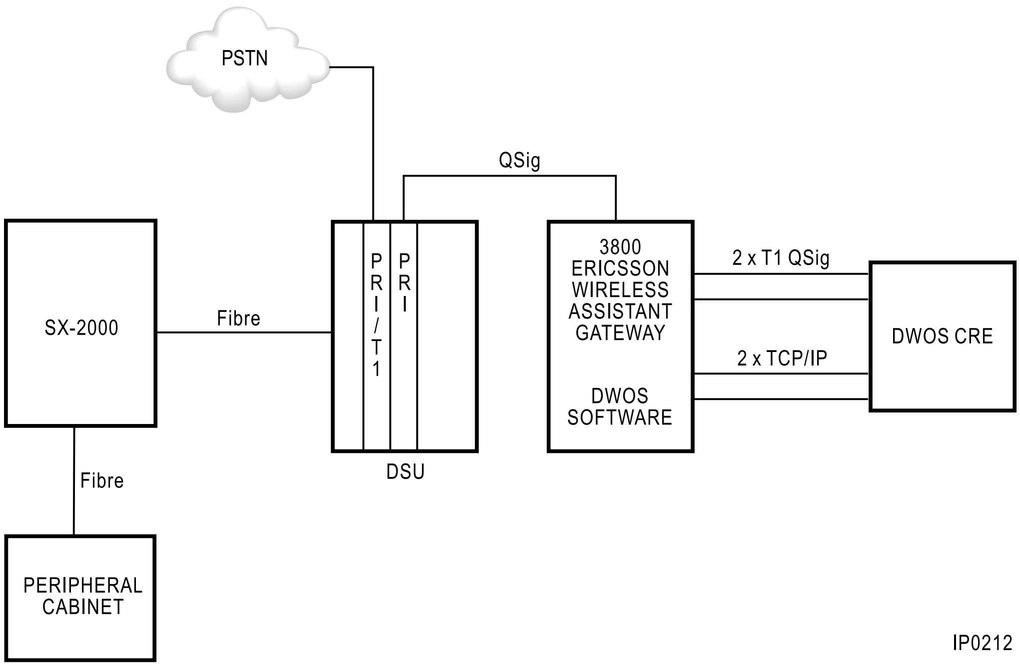
The Cellular Radio Exchange is directed by the 3800 Ericsson Wireless Assistant Gateway system and provides an interface between the Mobile Advantage Wireless Assistant and the wireline system. The Wireline system connects to the external PSTN through a PBX such as the Mitel® SX-2000®, Mitel SX-200 or March Networks 3300 Integrated Communications Platform.

Support for the following interfaces to wireline networks allows both Customer Premise Equipment (CPE) and Centrex scenarios to be supported:

- National ISDN-2 PRI
- National ISDN 2 (NI-2) 5ESS-2000 Switch PRI
- ECMA QSIG edition 2 on E1 and T1 facilities
- ETSI/DSS1 on E1

Radio Frequency Distribution (i.e. Macro Wireless Coverage)

This product uses existing Macro Wireless coverage to provide service to the wireless handsets. There is no additional Radio Frequency infrastructure included.



3800 ERICSSON WIRELESS ASSISTANT GATEWAY SYSTEM CONFIGURATION

DNIC Set Restrictions in the 3800 Ericsson Wireless Assistant Gateway system

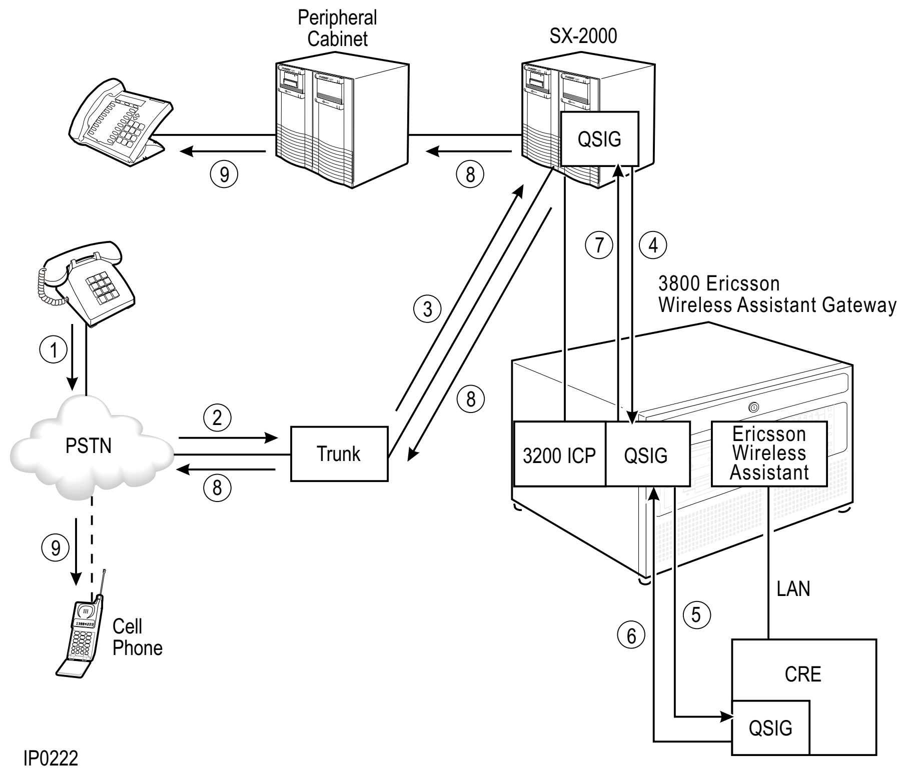
When cell phones are used together with DNIC phones in the 3800 Ericsson Wireless Assistant Gateway system, there is reduced functionality in the DNIC desktop set. This is a static and permanent condition within the system.

Basic Call Configuration for the 3800 Ericsson Wireless Assistant Gateway system

There are a number of scenarios that can occur within the 3800 Ericsson Wireless Assistant Gateway system context. For example:

- A call through the PSTN to 3800 Ericsson Wireless Assistant Gateway system (DNIC phone to DNIC phone with concurrent ringing of cell phone).
- A cell call through the Public Land Mobile Network (PLMN) to 3800 Ericsson Mobile Advantage Gateway system (Cell phone to DNIC phone with concurrent ringing of cell phone).
- An internal cell call from the 3800 Ericsson Wireless Assistant Gateway system to a cell phone in the PLMN.

The illustration below shows the pathway of a call from the PSTN to the 3800 Ericsson Wireless Assistant Gateway system with concurrent ringing enabled (desktop and cell phone):



The call pathway is explained in this manner:

1. A call is placed to the Public Switched Telephone Network (PSTN).
2. The PSTN takes the call and sends the call to a Trunk.
3. The Trunk routes the call to the SX 2000.
4. The call is then sent to 3800 Ericsson Wireless Assistant Gateway's QSIG.
5. The call is sent from 3800 Ericsson Wireless Assistant Gateway system QSIG to Wireless Assistant QSIG on the CRE.
6. The Wireless Assistant CRE then initiates two calls back over the QSIG to the 3800 Ericsson Wireless Assistant Gateway.
7. Both calls are sent from the 3800 Ericsson Wireless Assistant Gateway QSIG to the SX 2000 QSIG.
8. The one call is sent to the Peripheral Cabinet and the other to the PSTN.
9. The Peripheral Cabinet rings the desk phone and the PSTN rings the cell phone.

More information on the features and functions of the Ericsson Wireless Assistant can be found in the Wireless Assistant System Installation & Configuration Manual and the Wireless Assistant System Administration Manual.

WIRELESS SOFTWARE OPTION

The Wireless Software Option allows Symbol Technologies' proprietary, wireless H.323-based IP devices to be connected directly to the 3200 ICP system.

Note: The functionality of this software option is also offered as a adjunct server known as the 3800 Wireless Applications Gateway system. Note that this software option and the 3800 Wireless Applications Gateway system are mutually exclusive. Your system can have one of the two functionalities, but not both.

Advantages

The Wireless Software Option:

- Integrates voice and data on a single wireless LAN
- Provides mobile workers using wireless devices with PBX telephone features (see Features)
- Complies with the ITU H.323 standard, which
  - adheres to a Codec standard for compression and decompression of audio and data streams
  - ensures inter-operability between different equipment vendors
  - is network, platform, and application independent
  - provides conferencing (to allow three or more NetVision phones to communicate together).
- Extends voice features of the 3200 ICP system to Wireless LAN endpoints (Spectrum NetVision Phone and NetVision Data Phone)
- Can be managed from a remote location using the OPS Manager web-based management tool
- Provides conversion between Internet Protocol (IP) packet-based and Time Division Multiplexed (TDM) voice streams
- Supports two Spectrum24 Access points, a 2.0 Mbps Frequency Hopping Spread Spectrum (FHSS) Access Point and an 11 Mbps Direct Sequence Spread Spectrum (DSSS) Access Point. These Access Points (also called base stations) allow the deployment of several applications over the same infrastructure using a wide range of voice and data appliances.

The Wireless Applications Network

The Wireless Software Option extends wired voice and data systems to a converged wireless network.The wireless network consists of

- Access Points
- Wireless Endpoints
- 3200 ICP system Wireless Software Option

Access Points

The Spectrum24® Ethernet access points (APs), manufactured by Symbol Technologies, function as a Media Access Control (MAC) bridge to provide a transparent connection between wireless endpoints (NetVision telephony devices) and the wired Ethernet LAN.

The system supports two types of Spectrum 24 Access Points. The Spectrum24 AP-3020 Access Point is the Frequency Hopping Spread Spectrum (FHSS) Access Point used for in-building cellular networks. This Access Point operates at up to 2 Mbps. The Spectrum24 AP-4121 Access Point is the Direct Sequence Spread Spectrum Access Point used for in-building cellular networks. This Access Point operates at up to 11 Mbps. Both devices operate in the 2.4 to 2.5 GHz range and integrate seamlessly with wired environments.

Multiple APs can be deployed to create an integrated network that supports seamless, instantaneous roaming. The Spectrum24 AP-3020 FH Access Point conforms to the IEEE 802.11 standard for wireless LANs (WLANs), while the Spectrum24 AP-4121 DS Access Point conforms to the IEEE 802.11(b) standard. Both devices conform to the ITU H.323 standard for multimedia (voice, data, and video) communication.

Note: Although FH and DS networks may coexist, these devices are not interoperable.

Wireless Endpoints

The Wireless Software Option supports two wireless endpoints, the NetVision Phone and the NetVision Data Phone, both of which are manufactured by Symbol Technologies. FH Access Points support the 2 Mbps variants of these devices, and the DS Access Points support the 11 Mbps variants of these devices.

The wireless endpoints employ the ITU H.323 standard to convert analog voice signals into compressed digital packets. Using the TCP/IP protocol, the packets are transmitted to and from the Spectrum24 Access Point.

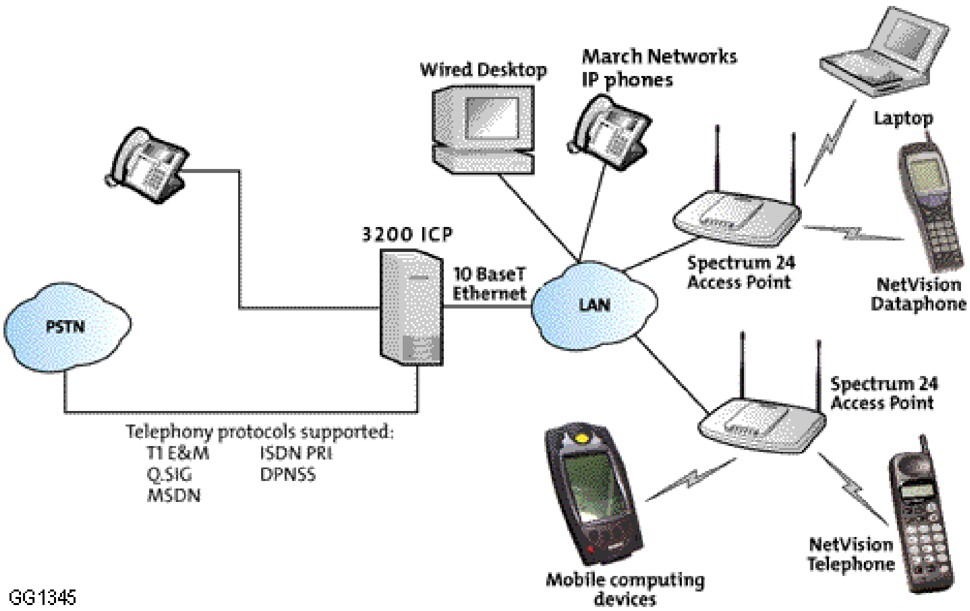
3200 ICP System Wireless Software Option

The Wireless Software Option provides the NetVision Phone and Data Phone with the features of an office desktop phone.

This software option completes protocol conversion between the PCM-based voice stream and IP packets, acts as a Registration Administration and Status (RAS) server to permit H.323 viability and control of features, and translates endpoint device IP addresses into system telephone extensions.

CONFIGURATION

The Wireless Software Option is an optional software package available on the 3200 ICP system.



CONFIGURATION WITH THE WIRELESS SOFTWARE OPTION ENABLED ON THE 3200 ICP SYSTEM

Note: Similar functionality is also available as an adjunct server, the 3800 Wireless Applications Gateway system. This Gateway connects to an existing PBX, known as a "host". Host PBXs from a variety of manufacturers are supported; however, the 3800 Wireless Applications Gateway system is optimized for connection to SX-2000® PBXs. Note that this software option on the 3200 ICP and the 3800 Wireless Applications Gateway system are mutually exclusive. Your system can have one of the two, but not both.

Traffic Dependent Configurations

Due to the unique nature of interfacing any wireless LAN configuration with the 3200 ICP system, it is recommended that you consult the Professional Services Division of Mitel Networks Corporation. They will assist you to determine your network's current compatibility with the this option. In some cases, modifications or upgrades to your network may be required.

PERFORMANCE CAPABILITIES

The Wireless Software Option has the following performance capabilities:

2 Mbps NetVision Phones

The 2Mbps NetVision Phones have the following capabilities:

- Supports G.711 (uncompressed voice) which allows 2 simultaneous conversations (2 speakers each) per Access Point.
- Due to the limitations imposed by G.711, the system supports a practical limit of 32 Symbol telephony devices.

Note: Up to 208 IP Devices can be supported on the 3800 Wireless Applications Gateway system.

- Provides a call rate of up to 1400 calls per hour.
- Provides up to 4 parties per conference with a maximum of 24 users on the conference bridge and a maximum of 5 simultaneous conferences at any one time.
- Emulates the Mitel Telephony User Interface (TUI) for program features.

11 Mbps NetVision Phones

For information on the performance capabilities of the 11 Mbps Direct Sequence Spread Spectrum networks, refer to traffic and performance guidelines provided by Symbol (www.symbol.com).

Key Protocols

The Wireless Software Option and the Spectrum24 access points use the IEEE 802.11 for FH networks and IEEE 802.11(b) for DS networks. They also comply with ITU H.323 protocols. IEEE 802.11 is used for wireless LANs, and ITU H.323 is used for multimedia (voice, data, and video) communication.

About IEEE 802.11

The IEEE 802.11 specification is a wireless LAN standard developed by the IEEE (Institute of Electrical and Electronic Engineering) committee that specifies an "over the air" interface between wireless endpoints, and between the wireless endpoints themselves. It also specifies wired local area networks. The Wireless Software Option employs IEEE 802.11 in an "infrastructure network" (as opposed to "ad hoc network") that uses fixed access points (Symbol's Spectrum24) to handle the transmission of data from the wireless to the wired medium. Within an infrastructure network, the access points and the wireless endpoints (Symbol's NetVision Phones and Data Phones) work together to define a coverage area known as a Basic Service Set, or cell. Wireless endpoints are free to roam from cell to cell, and if service areas overlap, handoffs can occur. This structure is very similar to the present day cellular networks around the world.

About ITU H.323

The H.323 standard provides a foundation for audio, video, and data communications across IP-based networks, including the Internet. By complying to this standard, multimedia products and applications from multiple vendors can operate together, allowing users to communicate without concern for compatibility. The H.323 standard will be the keystone for LAN-based products for consumer, business, entertainment, and professional applications.

The H.323 communications protocol serves as the "umbrella" set of standards that defines real-time multimedia communications and conferencing over packet-based networks. These standards define how components are built to comply with the H.323 standard, for example:

- set up calls, exchange compressed audio and video
- participate in multi-unit conferences
- operate with non-H.323 endpoints.

The H.323 protocol is the ITU-T Standard for Multimedia Conferencing on Local Area Networks. It is the standard for real-time communication over the Internet and corporate intranets.H.323 is part of a larger series of communications standards that enable video conferencing across a range of networks. Known as H.32X, this series includes H.320 and H.324, which address ISDN and PSTN communications.

MANAGEMENT TOOLS

The 3200 ICP system and the Symbol peripheral devices (Spectrum24 Access Points, NetVision Phones and NetVision Data Phones) have management tools that allow you to perform administration and maintenance.

- The 3200 ICP system offers the Java-based OPS Manager management tool
- The Spectrum24 offers the text-based Spectrum24 maintenance tool
- The NetVision Phones and Data Phones offer the Windows™-based NetVision Ipera™ 2000 Phone Administrator.

OPS Manager

OPS Manager is a complete telecommunications management tool that enables you to control the maintenance and operation of a network of elements. It is installed as an application on the 3200 ICP system. From the OPS Manager station, you can perform the following functions:

- add and delete users (move and change functionality for Symbol telephony devices is not supported in the OPS Manager Moves, Adds, & Changes application).
- schedule automatic upgrades, database saves, and database restores
- audit the server and Spectrum24 access points for alarms
- perform remote programming and maintenance
- program the NetVision phones
- locate unused directory numbers and unused virtual circuits on the gateway server.

Note: OPS Manager is a Java™-based application that supports multiple client stations. Therefore, you can access the application through a Netscape® Communicator 4.05 browser or a Microsoft™ Internet Explorer 4.01 browser from any Windows or Windows NT workstation on the network.

Note: The 3200 ICP system does not provide "Directory Server" support and for this reason you cannot integrate the telephone directory with a directory service database.

Spectrum24 Text-based Maintenance Tool

A text-based maintenance tool is embedded within each Spectrum24 Access Point. You can use it to set configuration options, download new firmware, and provide statistical displays. Use the following methods to access the maintenance tool's user interface:

- Telnet, using a wired or wireless Ethernet connection (the remote station requires a TCP/IP stack)
- Direct, using a serial connection (the remote station requires a communication program such as Hyperterminal)

- Dial-up, using a Hayes-compatible modem running at greater then 28,800 baud (the remote station requires a communication program such as Hyperterminal)
- Web browser, using an Internet connection (the remote station requires a TCP/IP stack, and the web browser must include JavaScript).

Note: Be sure to assign each AP an individual IP address.

More information on Spectrum24 access point management can be found in the Spectrum24 Access Point AP-3020 Product Reference Guide and in the Spectrum24 Access Point AP-4121 Product Reference Guide.

NetVision Ipera 2000 Phone Administrator

The NetVision Ipera 2000 Phone Administrator is a Windows 95/98/NT 4.0 application that is provided on the March Networks 3200 ICP system Software CD-ROM. It allows the system administrator to:

- create NetVision phone user data
- edit user data
- download user data to the NetVision phones
- update NetVision phone firmware

The system also includes a special PC serial cable to transfer data to the phones.

WIRELESS SOFTWARE OPTION PERIPHERALS

The Wireless Software Option supports two types of Symbol Spectrum24 Access Points, each of which supports a version of the Symbol NetVision Phone and NetVision Data Phone.

Note: Refer to the Symbol Web site ar www.symbol.com under Wireless Products for the latest information on the NetVision Phones, NetVision Data Phones and Spectrum24 Access Points.

Spectrum24 Access Points

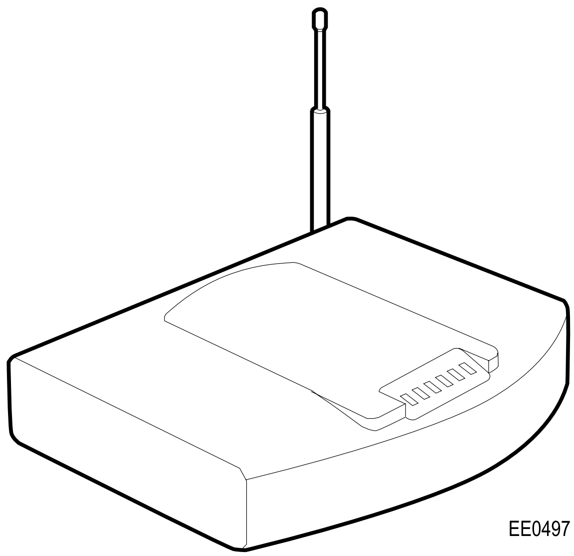
Spectrum24 Access Points (APs) act as a bridge between wired LANs and Spectrum24 wireless networks. They provide transparent access between the LAN and allow the wireless Symbol telephony devices to interface with the system.

The system supports two Spectrum 24 Access Points. The Spectrum24 AP-3020 Access Point is the Frequency Hopping Spread Spectrum (FHSS) Access Point used for in-building cellular networks. This Access Point operates at up to 2 Mbps. The Spectrum24 AP-4121 Access Point is the Direct Sequence Spread Spectrum Access Point used for in-building cellular networks. This Access Point operates at up to 11 Mbps. Both devices operate in the 2.4 to 2.5 GHz range and integrate seamlessly with wired environments.

The Spectrum24 has the following features and functionality:

- Built-in diagnostic capabilities with a power-up self-check
- A 4-way bridging architecture (wireless, Ethernet, PPP, internal stack)
- Wireless Media Access Control (MAC) interface
- 10 baseT Ethernet port interface with full-speed filtering

- 100 mW and 500 mW radio versions
- Power supply IEC connector and a country-specific AC power cable
- PC/AT Serial Interface
- Built-in antenna diversity
- Multiple antenna options
- Support for 127 Symbol telephony devices
- Simple Network Management Protocol (SNMP) support
- Repeater functions



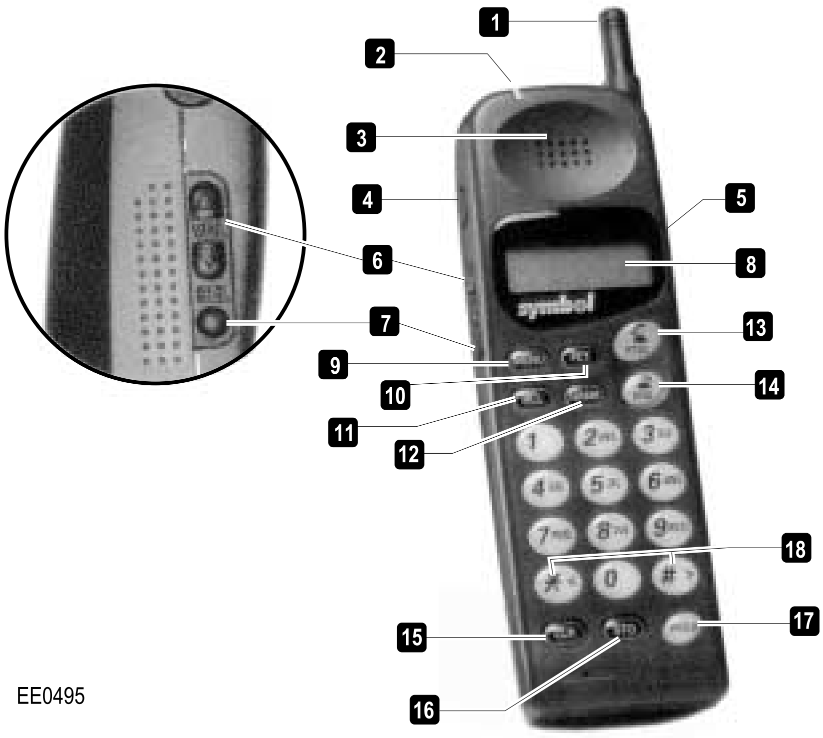
SPECTRUM24 ACCESS POINT

More information on Spectrum24 access point management can be found in the Spectrum24 Access Point AP-3020 Product Reference Guide and in the Spectrum24 Access Point AP-4121 Product Reference Guide.

NetVision Phones and Data Phones

The NetVision Phone and Data Phone are small, lightweight, wireless telephones designed to operate over Spectrum24 wireless data networks by using Voice-over-IP. There is a variant of the Phone and Data Phone designed for the AP-3020 and for the AP-4121.

By connecting the data network to the 3200 ICP system, you can use the NetVision Phone or Data Phone to make calls to other PBX extensions and to call destinations in the Public Switched Telephone Network (PSTN).



NetVision Phone

1. **Antenna** - provides signal reception (cannot be extended)
2. **LED** - indicates an incoming call
3. **Earpiece**
4. **Mini-headset jack** - allows connection of a headset for hands-free use
5. **Serial port** - allows software to be downloaded into the phone
6. **Volume adjust buttons** - adjusts the current call volume and scroll phone menus and lists
7. **SEL button** - selects a name from the name list
8. **LCD screen** - shows the phone status and menu options
9. **MENU key** - displays the phone user menu
10. **FCT key** - accesses the specially programmed features
11. **RCL key** - recalls the last number dialed, and accesses the speed-dial directory
12. **NAME key** - displays the names directory
13. **SND** key - outpulses the dialed number, answers an incoming call, powers on the phone and selects items in the phone menus
14. **END key** - ends a call, refuses a call, powers off the phone and returns to a previous menu
15. **CLR key** - removes the last digit entered from the keypad
16. **STO key** - stores a number in the speed-dial directory
17. **HOLD key** - places a call on hold

18. *< and #> keys - allows you to browse the menus and lists on the display screen.

NetVision Data Phone (EE0515)

1. **Antenna** - provides signal reception (cannot be extended)
2. **Earpiece**
3. **LCD screen** - shows the phone status, menu options, messages and applications prompts
4. **Serial port** - allows software to be downloaded into the phone
5. Scan key - Activates the scanner for data collection
6. **SND** key - outpulses the dialed number, answers an incoming call, powers on the phone and selects items in the phone menus
7. **HOLD key** - places a call on hold
8. **NAME key** - displays names from the call list
9. **RCL key** - recalls the last number dialed, and accesses the speed-dial directory
10. **MENU key** - displays the phone user menu
11. **STO key** - stores a number in the speed-dial directory
12. Microphone
13. **CLR key** - removes the last digit entered from the keypad
14. ***< and #>** keys - allows you to browse the menus and lists on the display screen.
15. **FCT key** - accesses the specially programmed features

16. **END key** - ends a call, refuses a call, powers off the phone, returns to a previous menu and ends data collection
17. Previous/Next keys
18. **Volume adjust buttons** - adjusts the current call volume and scroll phone menus and lists
19. **Mini-headset jack** - allows connection of a headset
20. **LED** - indicates an incoming call and a message waiting
21. Laser Scanner

Modes of Operation

- Personal mode
- Shared Login mode

Features of the Phone and Data Phone

- Speed dialing - NetVision telephony devices store phone numbers in 100 two-digit speed-dial locations. Users can store any dialed number.
- Caller ID - NetVision telephony devices display the IP address of the caller when an incoming call comes from a NetVision telephony device using peer-to-peer telephony.
- Optimized voice quality - NetVision telephony devices contain a voice-control mechanism that converts voice to digital data packets and back to voice to achieve high-quality audio.
- Rechargeable battery - NetVision telephony devices ship with a rechargeable Lithium-ion battery.
- Charging Cradle - The charging cradle is used to charge the Lithium-ion battery.

Refer to the NetVision Phone Quick Reference Guide or the NetVision Data Phone User Guide for more details on end-user operation.

FEATURES

This table lists the features that the Wireless Software Option can provide to NetVision Phone users:

FEATURES	
FEATURE NAME	**DESCRIPTION**
C D K M N O P R S T	
Call Pickup - Group	Call Pickup - Group lets you answer an incoming call that is ringing at another extension in your pickup group.
Call Pickup - Directed	Call Pickup - Directed lets you answer an incoming call that is ringing at another extension (PBX endpoint).
Call Park Retrieve	Call Park lets the attendant park a call so that a telephone user can remotely retrieve the call. You can retrieve parked calls from an Symbol telephony device.
	Page 1 of 3

FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Conferencing	Conferencing lets you connect three or more people into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.
ACD Login	The user engaged in Personal Login or Shared Login mode must be pre-programmed in the system as an ACD agent. In this mode, the ACD agent can use other call options such as Make Agent Busy.
Keep Alive	If you walk out of range of an access point, the NetVision telephony device will give visual and audio indications that it is out of the coverage area. When an NetVision telephony device is out of range of an access point, you are unable to make or receive calls. The Keep Alive timer specifies the length of time that the system should maintain state awareness of an NetVision telephony device that has gone out of range. If you get back into range before the Keep Alive timer expires, the call is re-established (maintained).
Message Indication	Your NetVision telephony device indicates whether you have messages waiting in your voicemail box.
Night Service	You can answer a call at night (night bell) from a NetVision telephony device if night service is activated on the 3200 ICP system. Press the FCT key.
Out of Range - Recovery	If the user who has moved out or range of an Access Point moves back into the coverage area before the NetVision telephony device Keep Alive Timer expires, the NetVision telephony device immediately provides a keep alive indication to the system. The NetVision telephony device then removes the visual and audio indications for outside a coverage area.
Out of Range Notification	When the phone cannot maintain an association with the network, it displays "No Network" in the status line and: • If the phone is idle, sounds the out-of-range tone (one long beep and two short beeps). Default tones can be set by the user. • If the phone is active, sounds the call-waiting tone (two short beeps) to the user. The user has 10 seconds to move within range of the network before the phone hangs up.
Out-of-Service Notification	If a user places a call to a NetVision phone that is out-of-service, the NetVision phone will display that the user with that particular set of IP/MAC codes is out of service.
	Page 2 of 3

Features (continued)	
Feature Name	**Description**
Personal Mode	This mode allows a NetVision telephony device to be assigned to an individual user. The NetVision telephony device automatically registers the user with the system when the NetVision telephony device is turned on. Once the NetVision telephony device registers with the system, the user can make and receive calls. In Personal Mode, one NetVision telephony device is assigned and associated with a single user.
Roaming	A NetVision phone can move between Access Points (APs) without losing connectivity, if the Media Access Codes are programmed into the respective phone. This is known as Roaming.
Shared Login Mode	This mode allows the system to support more NetVision telephony device users than there are physical sets in the system, based on the assumption that not all the users will require NVPs at the same time. In Shared Login Mode, any user can use any NetVision telephony device.
Speed Call - System	System Speed Call lets you dial stored system numbers.
System Re-routing	A call placed by a NetVision telephony device user can be re-routed due to call forward no answer, call forward busy, call forward always, or out of service. (Ensure the call is rerouted to a valid extension number.)
Transfer	Transfer lets you move a call from one telephone to another. Before completing a transfer, you can consult privately with the third party and swap between private conversations with each of the parties.
Unregistered Devices - Forward Call	If a registered NetVision telephony device user attempts to call to an unregistered device the call will be rejected. However, depending on system programming, the call can be forwarded to voicemail or to another directory number (DN).
	Page 3 of 3

Refer to Wireless Software Option Features for instructions on programming Wireless Applications Features.

IP Trunking

IP Trunking allows the 3200 ICP system to transport voice over IP networks.

The March Networks 3800 IP Trunking Package for the 3200 ICP system includes the following:

- March Networks 3800 IP Trunking software package
- IP Trunk Card (can be purchased in a 30 or 60 channel variant)

The IP Trunk behaves the same as an MSDN trunk except it uses the existing intranet to transport both voice and signaling data. The IP Trunk Card packetizes and streams voice to the appropriate network node. An MVIP cable must be attached to the IP Trunk Card which is connected to the other MVIP based cards.

The IP Trunk Card must be installed in slot 3 1 3.

Refer to Install IP Trunking and Program IP Trunking.

Note: For more information, an Online Book is provided on the March Networks 3800 IP Trunking software CD-ROM. Simply navigate to the Online Book folder and double-click the 3800_IP_Gateway.htm file.

ISDN Support

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

You can integrate your LAN traffic with your existing private or public digital network connections on Euro ISDN, DASSII (public access) protocols or even on your private DPNSS network. Flexibility is allowed for in the wide range of LAN protocols, notably Novell, Microsoft, IBM, Unix, ICL, DECLAT and Banyan - Vines.

Mitel's family of converged ISDN remote access solutions extends the power of the head office LAN and PBX to teleworkers and those in branch offices. See also XpressOffice 5232i.

ISDN Connectivity

ISDN connectivity is another step towards a converged voice and data network. ISDN access lets customers leverage the advantages of ISDN network services for both voice and data applications, effectively improving performance and network resource management while controlling costs.

The March Networks 3200 ICP system supports multiple ISDN protocols and provides ISDN connectivity. The PBX connects with the ISDN public network and data devices (i.e., routers, video conferencing equipment, servers, etc.) by using Primary Rate Interface (PRI). ISDN takes advantage of the following PBX features to capture and control costs, analyze peak periods, and fine tune network resources accordingly for both voice and data calls:

- ARS/LCR (Automatic Route Selection / Least Cost Routing)
- SMDR (Station Message Detail Recording)
- Min/Max Traffic Control
- Per Call Service Selection
- Limited Toll Restriction
- PBX Trunk Diagnostics
- NFAS (Non-Facility Associated Signaling)
- Remote LAN access

ISDN Primary Rate Interface

ISDN Primary Rate Interface (PRI) is becoming the most cost-effective enterprise solution for IT managers responding to increased demands for remote LAN access, Internet and Intranet access, off-site desktop and group video conferencing, and a host of other inbound and outbound data applications.

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

At the same time, ISDN supports enhanced voice communications capabilities. These include: Caller Line Identification Delivery (CLID) and Automatic Number Identification and Dialed Number Identification Service (ANI/DNIS), which allow you to know who's calling and facilitate call center and CTI applications; plus, fast call set-up, call-by-call, and Min/Max for reduced trunking. Of course, ISDN also delivers the highest degree of voice clarity of any transmission medium available.

ISDN Options

This section describes the purchasable options supported by the Dual T1 and Dual E1 Cards.

Min/Max controls the number of simultaneous incoming and outgoing calls. The level of control ranges from generic minimums and maximums on all calls to minimums and maximums for particular directory numbers.

Automated Min/Max works in conjunction with Min/Max, and increases Min/Max configurations by providing time of day programming. Time of day programming allows you to have consistent traffic control without having to frequently reprogram Min/Max. You can program call control for an entire week, and the system will automatically change Min/Max settings based on the time of day and the day of the week.

NFAS (Non-Facilities Associated Signaling) allows you to use a single D-channel to handle the signaling requirements for a group of PRI links that all use the same Protocol. This feature eliminates the need to purchase a D-channel for each link. NFAS is mainly for North America.

D-channel Backup provides an alternate D-channel for calls related to NFAS. If the active D-channel fails, the system switches to the backup D-channel to support call processing. This functionality is mainly for North America.

Remote LAN Access lets you configure the card as if it was the interface from the Public Network. It will allow you to send and receive information to another ISDN compatible device, such as a router, server or another PBX. This feature is mainly used for e-mail and Internet access.

Conditions

- Min/Max is required in order to program Auto Min/Max.
- NFAS is required in order to program D-Channel Backup.

Programming

Programming is performed through IMAT (ISDN Maintenance and Administration Tool). Refer to Installing IMAT for the installation instructions.

OPS MANAGER

OPS Manager is a complete telecommunications management tool that enables you to control the maintenance and operation of a network of Mitel PBXs. See OPS Manager for detailed information to install, program, maintain and troubleshoot OPS Manager. From the OPS Manager station, you can perform the following functions on a PBX or on network of PBXs:

- manage a network telephone directory
- move, add, change, and delete users
- integrate the network telephone directory with a directory service database
- schedule automatic upgrades (on LIGHT PBXs in the network), database saves, and database restores
- monitor alarm status messages that are automatically reported from the network
- audit the network elements for alarms
- perform remote network element programming and maintenance
- locate unused directory numbers and unused circuits.

The OPS Manager application is available

- as software only; that is, you can install the application on your own server
- as a turn-key platform; that is, a server with the required hardware and software installed.
- as an application on the March Networks 3200 ICP system server.

Note: OPS Manager is a Java™-based application that supports multiple client stations. Therefore, you can access the application through a Netscape® Communicator 4.05 browser or a Microsoft Internet Explorer browser from any Windows NT or Windows 95 workstation on the network.

NUPOINT MESSENGER

NuPoint Messenger is a PC-based voicemail and messaging package that works with the Mitel PBX. It provides online maintenance and configuration, and enhanced security. NuPoint Messenger can support up to four languages per system simultaneously. It also supports individual, workgroup, and enterprise requirements, softkey integration on SUPERSET™ telephones, and allows for centralized voicemail for organizations with multiple PBXs.

NuPoint Messenger modules offer the following:

- Call Processing and Auto Attendant
- Integrated Fax Messaging and Management
- Paging Support
- Lodging Industry Support
- PC-based Client Applications for Mailbox Control.

XpressOffice 5232i

XpressOffice 5232i brings office functionality to the comfort of your home. It connects your two most important resources, your computer and your SUPERSET digital telephone, directly to your corporation with a high-speed ISDN link. This connection allows you to access the corporate LAN and use your telephone just as you would at the office.

There are two network connection options for the XpressOffice 5232i: as a bridge or a router. The option that you choose depends on your corporate network setup and the filtering options that you want to apply to the data traffic. The telephone connection is simpler. You configure a direct connection to your office PBX, permitting your home-based SUPERSET telephone to function as an extension of your office telephone network.

The following option packs are available for the XpressOffice 5232i:

- Hub Pack
- Voice Pack (Analog)

Note: Option packs must be installed by a Mitel representative.

Configuration

Servers - Minimum Configurations

Different servers are provided for different configurations.

Base Server Hardware	
System	**Description**
3200 ICP system	TS800 Server PII 350 MHz, 256 Mb RAM
	TS1500SR Server PII 350 MHz, 256 Mb RAM
	1400 PIII 850 MHz, 256 Mb RAM
	1400SR (UK only) PIII 850 MHz, 256 Mb RAM
3800 Ericsson Mobile Advantage Gateway system	TS1500SR PIII 550 MHz, 512 Mb RAM
3800 Ericsson Wireless Assistant Gateway system	TS800 (with Riser card) PIII 550 MHz, 384Mb RAM
3200 ICP system with Ericsson Wireless Assistant	TS1500SR PIII 550 MHz, 512 Mb RAM
3200 ICP system with Wireless Software Option	TS800 (with Riser card) PIII 550 MHz, 512 Mb RAM
	1400 PIII 850 MHz, 512 Mb RAM
3200 ICP system with IP Trunking	1400 PIII 850 MHz, 256 Mb RAM
	TS1400SR (UK only) PIII 850 MHz, 256 Mb RAM
	TS1500SR PII 350 MHz, 256 Mb RAM
3200 ICP system with Wireless Software Option and IP Trunking	1400 PIII 850 MHz, 512 Mb RAM
	1400SR (UK only) PIII 850 MHz, 512 Mb RAM
	TS1500SR PIII 550 MHz, 512 Mb RAM

BASE SERVER HARDWARE

The 3200 ICP system server is shipped with the following installed hardware components:

- Industry-standard computer with a mouse and keyboard
- 4-port serial card (optional for the 1400 and TS1500SR only);
- 10 Mbps or 100 Mbps Ethernet NIC
- Main Fiber Controller (MFC) card that connects the peripheral cabinet
- Tone and Conference Digital Signal Processor (DSP) card
- Dual T1 or Dual E1 cards that allow you to connect digital links (maximum of two cards with two links per card)
- one E2T card for Real Time Complex (RTC) functionality
- a second ET2 Card for IP to TDM functionality

3800 Ericsson Mobile Advantage Gateway system/3800 Ericsson Wireless Assistant Gateway system/3200 ICP system with Ericsson Wireless Assistant

Base server hardware, plus:

- Adaptec four-port NIC

3200 ICP system with IP Trunking

- IP Trunk Card TP240-30 for 30 channels or TP240-60 for 60 channels

Refer to the card layouts of the TS1500, 1400 and TS800.

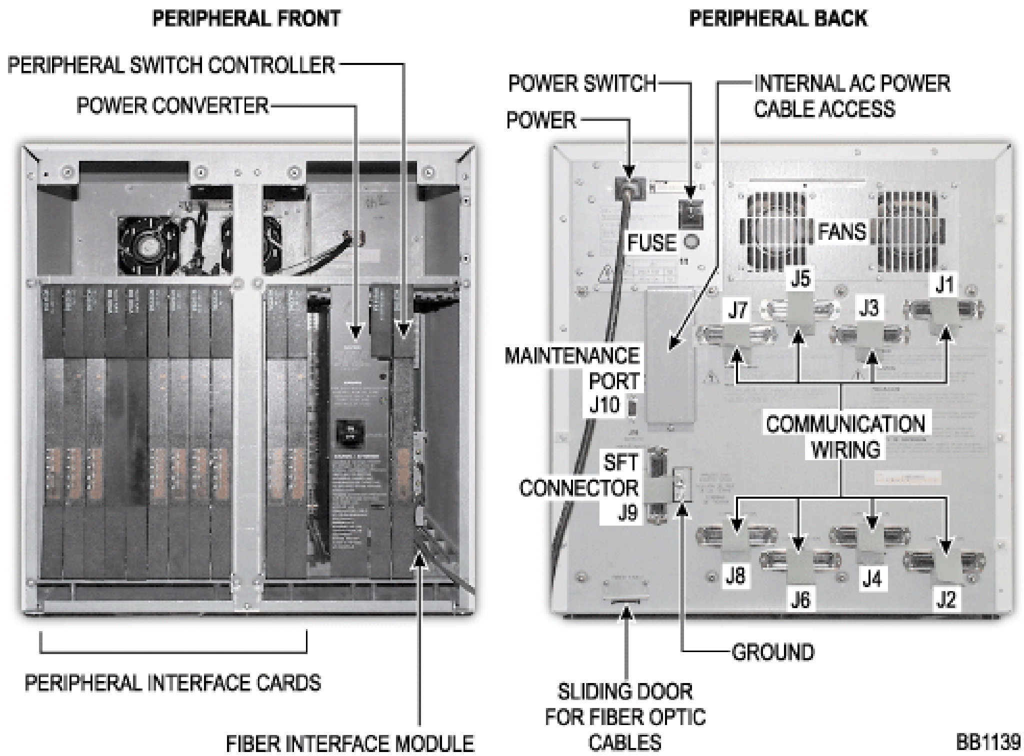
SERVER SOFTWARE

The 3200 ICP system is shipped with the following software components:

- 3200 ICP system software
- OPS Manager software
- Mitel Telephony Application Interface (MiTAI),
- ISDN for Windows NT and IMAT (ISDN Maintenance and Administration Tool)
- Microsoft Windows NT 4.0 Server Operating System
- Microsoft Internet Information Server x (IISx) and Active Server Pages (ASP)
- Microsoft Windows NT Service Pack and Microsoft Internet Explorer
- NetVision Ipera 2000 Phone Administrator application
- Field Change Instruction Document.

PERIPHERAL CABINETS

Each peripheral cabinet holds up to 12 Peripheral Interface Cards and provides up to 192 ONS or DNI ports. By purchasing the Peripheral Node Expansion feature package, a slave cabinet can be added that expands the node up to a total of 384 ports and 24 peripheral interface cards (the number of voice channels remains the same). One Peripheral Switch Controller (PSC) card and one Fiber Interface Module (FIM) is installed in the master cabinet of each peripheral node. The PSC card provides control for all peripheral interface cards, and fiber optic cable connects the FIM to the main control.



Peripheral Cabinet II

The peripheral cabinet consists of the following components:

- **Peripheral Interface Cards:** The peripheral interface cards connect telephone trunks and peripheral devices (such as SUPERSET telephones) to the system. They are located in slots 1 through 12.
- **Power Converter (AC):** The AC power converter converts AC input power to the voltages required by the circuit cards and FIMs (+5 Vdc, +12 Vdc, -27 Vdc, -48 Vdc and 80 Vac ringing). It is installed in slots 13 to 15.
- **Power Converter (DC):** The DC power converter converts DC input power to the voltages required by the circuit cards and FIMs (+5 Vdc, +12 Vdc, -27Vdc, and 80 Vac ringing). It is installed in slots 13 to 15.
- **Peripheral Switch Controller card (PSC):** The PSC card performs all peripheral switch functions for up to 12 peripheral interface cards (or 24 cards with the addition of a peripheral slave cabinet, see Peripheral Node Expansion feature package for details). It is installed in slot 16 of the master peripheral cabinet.
- **Fiber Interface Module (FIM):** The FIM connects the peripheral node to the control node. It is installed in slot 17 of the master peripheral cabinet.
- **Cabinet Frame:** Each peripheral cabinet has 17 slots numbered from left to right. Slots 1 to 12 support peripheral interface cards and slots 13 to 15 hold the Power Converter. A master peripheral cabinet also holds a PSC card in slot 16, a FIM in slot

17, and a Peripheral Interconnect card in slot 16B (if your node is expanded). A peripheral slave cabinet holds a Peripheral Interconnect card in slot 16, in addition to the peripheral interface cards and Power Converter. Slots 16B and 17 of the slave cabinet are not CDE programmable (for more information on expanded peripheral nodes, see the Peripheral Node Expansion feature package).

- **Power Distribution Unit (PDU) (AC):** The AC PDU filters and switches the 120/240 Vac input power to the Power Converter and fan assembly.
- **Power Distribution Unit (PDU) (DC):** The DC PDU filters and switches the -48 Vdc input power to the Power Converter and fan assembly. Note that the server is available in AC version only.
- **Fan Assembly:** Two fans in the removable fan assembly cool the cabinet.
- **Rear Panel:** The following switches and connectors are located on the rear panel of the cabinet:
  - A power on/off switch
  - A fuse to protect the line lead on the input power (AC systems) or circuit breaker (DC systems)
  - A 3-conductor male receptacle to connect AC input power
  - A sliding door for the Tx and Rx fiber optic cables
  - An RS-232 maintenance terminal port for remote access (remote maintenance connections will only work on the master cabinet of a peripheral pair)
  - Nine 25-pair male, filtered, Amphenol connectors are located on the rear panel. All lines and trunks from the main distribution frame connect to the eight horizontally positioned connectors using 25-pair cable. The single vertically positioned 25-pair D-phone connector provides power and contact closure to an optional external system fail transfer unit.
  - A 3-conductor female plug is recessed in the rear panel behind a small cover plate (AC systems only). The plug connects to the power connector on the AC Power converter.
  - A ground connector.

Note: The March Networks 3200 ICP system does not support the RS-232 maintenance port on the peripheral node.

SUPERSET HUB

The SUPERSET HUB builds on the distributed PBX concept by delivering advanced digital telephony functionality for workgroups. It allows SUPERSET telephones to be cost-effectively provided where a distributed system would be too expensive.

The SUPERSET HUB provides DNIC connectivity through an RJ-45 patch panel and Fibre Interface Module (FIM) connection to the host peripheral cabinet. At the peripheral cabinet, a carrier module provides the connection for the FIM and interfaces back into the PBX. The SUPERSET HUB may be rack-mounted or wall-mounted.

SUPERSET HUB

SYSTEM FAIL TRANSFER

The SFT maintains telephone service in the event of system failure (such as a power outage). When the system goes into SFT mode, the SFT unit connects up to six internal POTS telephone extensions directly to the CO, bypassing the PBX completely.

The SFT is an optional, stand-alone, wall-mounted device that connects to the system's peripheral cabinet or main distribution frame (MDF). Each SFT can control six circuits, and up to four SFTs can be daisy-chained together for each zone, providing security for 24 internal extensions.

The SFT switches to SFT mode under the following conditions:

- Failure of the system power converter
- Failure of the system main control (in a redundant system, both main control planes must fail, causing a critical alarm to all zones)
- Interruption of the system AC power
- Failure of the peripheral switch controller (zone)
- Loss of the fiber link between the main control and peripheral cabinets.

Power Supply

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

Transfer Relays

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

Loop Detector

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

SFT Control Leads

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

Power Consumption

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

Power Dissipation (watts)

POWER SUPPLY	TYP. (WATTS)	TYP+20%
-48Vbat	3.18	3.81
@Vbat=-56 V	3.71	4.45

LAN/WAN NETWORK CONFIGURATION

To maintain optimum voice quality, it is recommended that voice and data traffic be segregated as much as possible. Refer to Before You Begin.

Methods to achieve this are:

- Run Voice and Data on separate physical networks
- Run Voice and Data on separate Virtual LANs (VLAN) with priority (IEEE standard 802.1p/Q)
- Use a separate subnet for voice traffic
- Use Ethernet Switches instead of hubs.
- Use Full Duplex Fast Ethernet
- Use Full Duplex Fast Ethernet and Ethernet Trunks between switches
- When IP phones are being placed across routed links then the routers should be configured to prioritize voice traffic based on Type of Service (TOS) using techniques such as Weighted Fair Queuing (WFQ) with multiple queues configured. For example: high priority for voice and low priority for data. On slow WAN links between routers set the Maximum Transmittable Unit (MTU) appropriately for the speed of the WAN link.

Peripherals

The system connects to any of the following peripheral devices:

Telephones

- SUPERSET 4001, page 41, single-line telephone
- SUPERSET 4015, page 42, SUPERSET 4025, page 43, SUPERSET 4125, page 44, and SUPERSET 4150, page 45, multiline telephones

IP Telephones

- SUPERSET 4015 IP, page 47, and SUPERSET 4025 IP, page 48, multiline telephones that connect directly to a 10BaseT Ethernet network
- 5010 IP, page 49, and 5020 IP, page 50, multiline dual-port telephones that connect directly to a 10BaseT Ethernet network.

Programmable Key Modules

- SUPERSET PKM12 Programmable Key Module, page 52
- SUPERSET PKM48 Programmable Key Module, page 53

Digital Line Monitors

- Single-line Digital Line Monitor, page 54

Attendant Consoles

- SUPERCONSOLE 1000®, page 55 attendant console
- SUPERSET 7000, page 56, attendant console

Datasets

- Dataset 2103, page 59
- Dataset 2203., page 60

Other Devices

- Console DSS/BLF Interface Unit, page 58
- Analog devices such as the SUPERSET Interface Module, page 51, ONS telephones, fax machines, or modems
- DNIC Music On Hold /Pager unit., page 61
- March Networks 5423 IrDA Module, page 62

Wireless Applications Software Option peripherals

Refer to Wireless Software Option Peripherals, page 21, for information on the Symbol peripherals.

Discontinued Peripheral Devices

MILINK Data Module

MILINK Programmable Key Module

TELEPHONES

The system supports the following SUPERSET 4000 series telephones:

- SUPERSET 4001, page 41 single-line telephone
- SUPERSET 4015, page 42 multiline telephone with basic LCD display
- SUPERSET 4025, page 43 multiline telephone with enhanced LCD display
- SUPERSET 4125, page 44, multiline telephone with enhanced LCD display and built-in RS-232 interface for computer connection
- SUPERSET 4150, page 45 multiline telephone with touch-sensitive LCD display and built-in RS-232 interface for computer connection.

The SUPERSET 4000 series telephones, SUPERSET PKM48 and PKM12 Programmable Key Module are available in light or dark grey.

Adding a SUPERSET Interface Module to a SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone lets you connect to additional devices.

SUPERSET INTERFACE MODULE	CONNECTED DEVICE
SIM1	PKM 12, page 52, PKM48, page 53
SIM2	analog devices
Note: You cannot connect a PKM48 to a SIM2.	

SUPERSET 4001 Telephone

The SUPERSET 4001 telephone is a single-line, digital telephone with

- Seven Speed Call keys
- Four fixed-function keys: Program, Hold, Flash, and Message
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4001 Telephone

SUPERSET 4015 TELEPHONE

The SUPERSET 4015 telephone is a multiline, digital telephone with

- Twenty-character alpha-numeric liquid crystal display (LCD)
- Seven line keys, each with a built-in line status indicator
- Six fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, and Message
- Automatic selection of prime line or ringing line
- Key selection of non-prime line
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4015 TELEPHONE

SUPERSET 4025 Telephone

The SUPERSET 4025 telephone is a multiline, digital telephone with

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

The SUPERSET 4025 supports the SUPERSET Interface Module, page 40 for connection to additional devices.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4025 Telephone

SUPERSET 4125 Telephone

The SUPERSET 4125 telephone is a multiline, digital telephone with

- Twenty-character alpha-numeric liquid crystal display (LCD) with contrast control
- Three softkeys for feature access
- Fourteen line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, and Speaker
- Built-in RS-232 interface for a computer connection
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation (half-duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

The SUPERSET 4125 supports the SUPERSET Interface Module, page 40 for connection to additional devices.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4125 TELEPHONE

ES0010

SUPERSET 4125 Telephone

SUPERSET 4150 Telephone

The SUPERSET 4150 telephone is a multiline, digital telephone with

- Forty-character alpha-numeric liquid crystal display (LCD) with contrast control and six touch-sensitive softkey areas for feature access
- Fourteen line keys, each with a built-in line status indicator
- Four fixed-function keys: SuperKey, Hold, Redial, Speaker, and Microphone
- Built-in RS-232 interface for a computer connection
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation (full-duplex if AC adapter is plugged in)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

The SUPERSET 4150 also accepts a SUPERSET Interface Module, page 40 that lets you connect to additional devices.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4150 Telephone

IP TELEPHONES

The system supports the following IP Telephones.

- SUPERSET 4015 IP, page 47, multiline telephone has a basic LCD display, and connects directly to a 10BaseT Ethernet network
- SUPERSET 4025 IP, page 48, multiline telephone has an enhanced LCD display, and connects directly to a 10BaseT Ethernet network
- March Networks 5010 IP, page 49, telephone multiline telephone has a basic LCD display, and connects directly to a 10BaseT Ethernet network.
- March Networks 5020 IP, page 50, telephone multiline telephone has an enhanced LCD display, connects directly to a 10BaseT Ethernet network and has two LAN ports.

The SUPERSET 4015 IP and 4025 IP telephones have been discontinued but are still supported.

SUPERSET 4015 IP TELEPHONE

The SUPERSET 4015 IP telephone is a multiline, digital telephone with

- Twenty-character alpha-numeric liquid crystal display (LCD)
- Seven line keys, each with a built-in line status indicator
- Six fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message
- Automatic selection of prime line or ringing line
- Key selection of non-prime line
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4015IP TELEPHONE

SUPERSET 4025 IP TELEPHONE

The SUPERSET 4025IP telephone is a multiline, digital telephone with

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

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET 4025IP TELEPHONE

March Networks 5010 IP Phone

The 5010 IP telephone is a multiline, digital telephone with

- Two LAN ports
- Twenty-character alpha-numeric liquid crystal display (LCD)
- Seven line keys, each with a built-in line status indicator
- Six fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message
- Automatic selection of prime line or ringing line
- Key selection of non-prime line
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

March Networks 5010 IP Phone

MARCH NETWORKS 5020 IP PHONE

The SUPERSET 5020 IP telephone is a multiline, digital telephone with

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

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

MARCH NETWORKS 5020 IP PHONE

SUPERSET Interface Module

The SUPERSET Interface Module 2 (SIM2) provides analog interface functionality that lets you connect one or more 2-wire analog devices (such as ONS telephones, fax machines, or modems) to the second B-channel of your SUPERSET 4025, SUPERSET 4125, or SUPERSET 4150 telephone. The sum of the Ringer Equivalence Numbers (REN) of all devices in the loop cannot exceed 2.0 REN. The analog device has its own directory number and operates independently from the host telephone; however, if you attach multiple analog devices in parallel, they share the second B-channel.

The analog device connects to the Analog/Fax/Modem interface on the SIM2, page 40.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERSET INTERFACE MODULE 2 (SIM 2)

ES0099

Programmable Key Modules

Note: PKMs are not supported for IP devices on the 3200 ICP system.

SUPERSET PKM12 Programmable Key Module

The SUPERSET PKM12 Programmable Key Module (PKM) is a digital device which provides 12 additional personal keys for SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones. Each personal key can be programmed as a Feature key, Direct Station Select (DSS) key, Speed Call key, or for other uses. Each key has a Line Status Indicator that behaves the same as the indicators on the SUPERSET 4000 series telephones.

The PKM12 connects to a SUPERSET 4000 series telephone using the included modular cable, and a SUPERSET Interface Module (SIM1) installed in the set. The module supplies power to the PKM12.

Note: The PKM12 and PKM48 are the only programmable key modules qualified by MITEL for connection to SUPERSET 4000 series telephones. The PKM12 is not designed to connect to a PKM12.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

Note: The PKM12 is programmed the same as a PKM48, but do not program keys 13 or above.

SUPERSET PKM12 Programmable Key Module

SUPERSET PKM48 PROGRAMMABLE KEY MODULE

The PKM48 Programmable Key Module provides 48 additional feature keys for SUPERSET 4025, SUPERSET 4125, and SUPERSET 4150 telephones. Each feature key has a Line Status Indicator that behaves the same way as those on SUPERSET 4000 series telephones.

The PKM48 connects to the PKM48 interface on the SIM1, page 40. You cannot connect a PKM48 to a SIM2, page 40.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

ES0100

DIGITAL LINE MONITOR

SINGLE-LINE DIGITAL LINE MONITOR

The Single-line Digital Line Monitor (DLM) is used to record voice information from Mitel DNIC-based digital telephones or consoles on an externally connected tape recorder (not included). The unit records calls to/from the associated telephone only. The unit can be located anywhere within the building, wall mounted, or placed under a SUPERSET telephone. No PBX programming is required.

A default warning tone is repeated on the line every 15 seconds, which can be heard by all parties in a monitored conversation and on the recording (the tone can be disabled). The DLM is transparent to the signals passing between the PBX and DNIC telephone, unless the warning tone is enabled.

When power is removed from the DLM, the unit is by-passed internally. The DNIC telephone or console will continue to work, but the conversation will no longer be recorded.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

CC0921

Attendant Consoles

SUPERCONSOLE 1000 Attendant Console (SUPERSET 6DN)

The SUPERCONSOLE 1000 attendant console is used to perform call handling functions as well as some maintenance and administrative functions (such as moves and changes). The 4-line by 80-character alphanumeric display shows source and destination information, time and date information, call waiting information, and station information (such as COS and COR values).

The console has

- Fourteen hardkeys
- Four programmable firmkeys (for access to purchased options such as Hotel/Motel)
- Ten softkeys
- A dial pad (for both alphabetic and numeric input)
- Volume controls
- Integral handset
- Connector for a headset
- An RS-232 serial printer port.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

SUPERCONSOLE 1000 Attendant Console

SUPERSET 7000 ATTENDANT CONSOLE

The SUPERSET 7000 attendant console consists of TALK TO® card, SUPERSET 400 series handset, and handset cradle connected to a computer. The TALK TO card uses a standard DNIC interface to communicate with the PBX. The SUPERSET 7000 attendant console connects to the system by using peripheral interface cards. These cards provide telephone trunk and telephone extension information to the console.

The console has

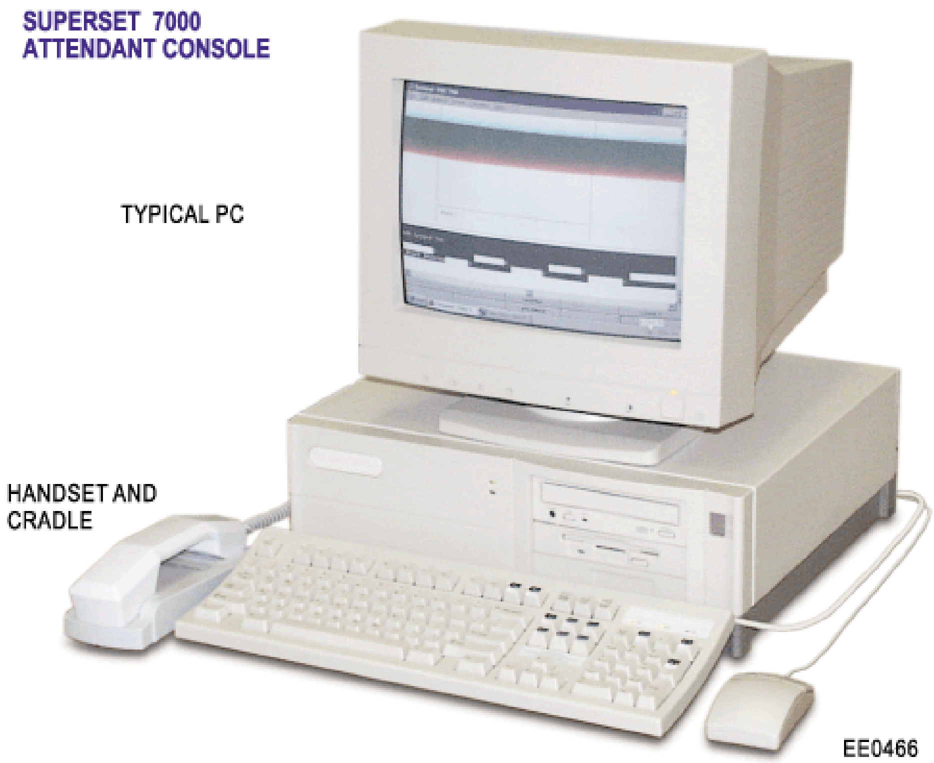
- Eight attendant function keys
- One hold key
- Three programmable firmkeys
- Ten softkeys
- A dial pad
- Volume and arrow keys
- Integral handset.

The SUPERSET 7000 application runs on a customer-supplied computer that meets the following minimum requirements:

- 80486/66 MHz processor with 16MB of RAM and Windows™ 3.1. (A Pentium processor with Windows 95 and 16MB of RAM are recommended.)
- VGA monitor
- 3fh inch floppy drive
- An AT 101 enhanced keyboard
- An ISA slot for the TALK TO card
- A Sound card is optional; having one allows you to adjust the console ringer volume and cadence.

Note: The SUPERSET 7000 application will not run under Windows NT and Windows 98.

Note: For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.



SUPERSET 7000 Attendant Console

CONSOLE DSS/BLF INTERFACE UNIT

The Console DSS/BLF Interface Unit uses a separate line connection to a DNIC port. The PKM48 attaches to the Console DSS/BLF Interface Unit and is associated with the attendant console through Customer Data Entry (CDE). The maximum cable length between the unit and the PKM48 is 5m (16.4 ft).

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.

DATASETS

Datasets provide data communication facilities for terminals, computer ports, and other types of data circuits that are switched through the system. These datasets are the interface between the Digital Network Interface Circuit (DNIC) in the PBX and the data devices connected to the system. The datasets transmit data and control signals over a single twisted pair of wires. Using a Mitel Dataset will allow you to carry voice and data communication over the same line.

The system supports the following datasets:

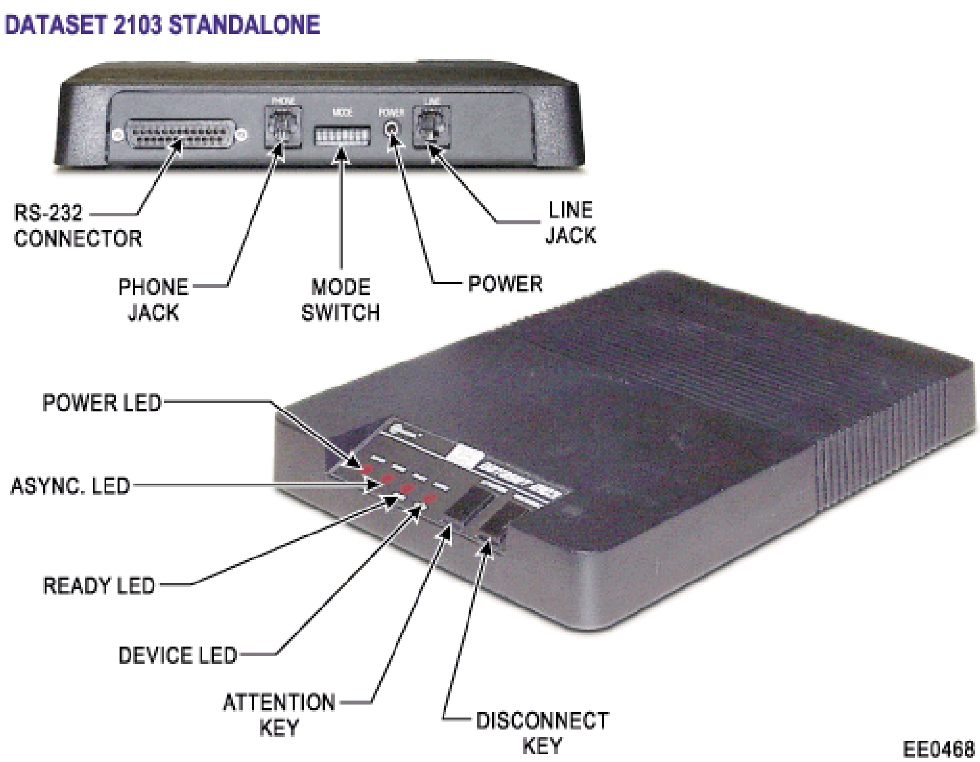
- DATASET 2103, page 59,
- DATASET 2203, page 60,
- MILINK Data Module (North America only)

DATASET 2103

The stand-alone DATASET 2103 is a synchronous/asynchronous dataset which is used to connect peripheral data devices to the PBX.

The DATASET 2103 is available in black only.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.



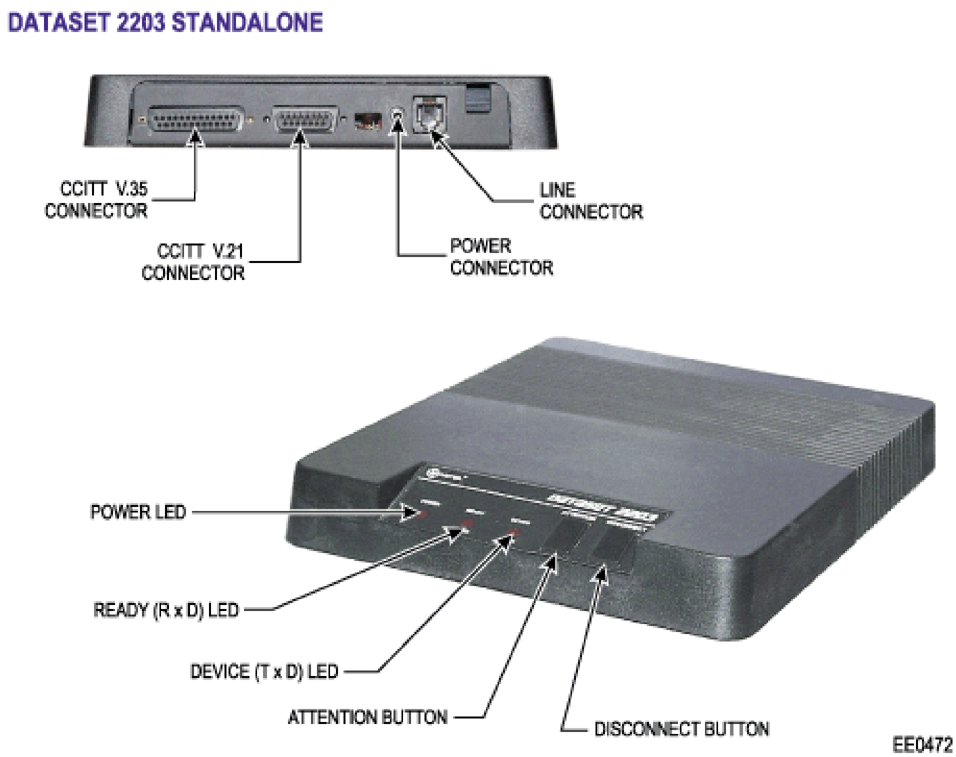
DATASET 2103

DATASET 2203

The stand-alone DATASET 2203 is a synchronous dataset which provides transparent synchronous communications for X.21 and V.35 compatible terminals.

The DATASET 2303 is available in black only.

For installation instructions, see the Install Peripherals section of the Technical Documentation CD-ROM.



DATASET 2203

DNIC Music On Hold/Pager Unit (DMP)

The DNIC Music On Hold/Pager (DMP) unit interfaces a port on the DNI Line card to

- an external music source for Music On Hold
- an external paging amplifier for Paging.

The unit is powered by the PBX and does not require a separate power source. A single 25-pair amphenol cable connects to the PBX via the main distribution frame (MDF); the unit can be wall-mounted. A single LED indicator provides basic status information.

Each DMP supports one music source and one paging zone, and the PBX supports one music source and 16 paging zones (15 individual zones and 1 "all" zone). If you need more than one paging zone, you can use multiple units; for example, for 16 paging zones, you need 16 DMP units.

Note: You can combine E&M paging with DMP Paging.

For installation instructions, see the Install System section of the Technical Documentation CD-ROM.

Note: "Use of the "Music on Hold" feature may under applicable copyright or other provincial, local, state and/or federal rules, regulations and/or statutes require that you obtain a licence from the local performing rights society or copyright owner, before you can provide music on hold to telephone users. Contact your music supplier for more information."

DNIC Music On Hold / Pager Unit

MARCH NETWORKS 5423 IRDA MODULE

The 5423 IrDA Module enables wireless IP-based desktop applications on the SUPERSET 4025 IP Phone or 5020 IP Phone.

Once installed on the an IP Phone, the 5423 IrDA Module allows users to make and receive calls using wireless IP-based desktop applications such as the Palm PDA. By pointing a PDA at his or her IP phone's infrared IrDA Portal, a user can dial the phone.

Users can also transform any SUPERSET 4025 IP Phone or 5020 IP Phone in their business into their own extension.

MARCH NETWORKS 5423 IRDA MODULE

Features

Standard Features

The following table lists the standard features available on the PBX.

Feature Name	Description
Standard Features	
Account Codes - Default	Default Account Codes are entered automatically by the system each time a user dials an external number. They may be used to segregate groups in SMDR for billing.
Account Codes - Verified and Non-Verified	Verified Account Codes let you access features that are not normally available at a station. These Account Codes can be used at any station to change the COS and COR. Non-Verified Account Codes let you enter codes on the SMDR record for billing and/or call management.
Account Codes - System	System Account Codes are automatically outpulsed by the system when outgoing calls are made on a specialized carrier trunk circuit.
Add Held	Add Held lets you move a call on Hold to another line appearance, form a conference with a call on Hold, or add a call on Hold to an existing conference.
Advice of Charge	Advice of Charge (AOC) allows the caller to determine the cost of a toll call. The AOC supports calls on the March Networks 3200 ICP system E1 card.
Attendant Access	See Attendant Directory Number, page 64.
Attendant Alarm Indications	See Attendant Console Status Display, page 64.
Attendant Busy-Out (Console)	Attendant Busy-Out (Console) places your attendant console in a busy-out condition (absent status) under certain circumstances. In the busy-out condition, incoming calls are automatically rerouted.
Attendant Busy-Out (Station)	Attendant Busy-Out (Station) lets you busy-out a specific station by using the attendant console. When you busy-out the station, it cannot be used or accessed.
Attendant Call Information Display	The Attendant Call Information Display provides the attendant with information about called and calling parties.
Attendant Call Selection	Attendant Call Selection lets you choose which group of incoming calls to answer first; each group is selected by pressing a softkey on the attendant console.
	Page 1 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Attendant Conference	Attendant Conference lets the attendant set up one or more conference connections between central office trunks and internal stations.
Attendant Consoles	See Peripherals, page 39.
Attendant Consoles (Multiple)	Multiple Attendant Consoles can be supported by the PBX. The number of consoles required must be determined by the customer prior to system installation.
Attendant Console Firmkeys	Attendant Console Firmkeys on your console can be programmed as one of the following feature keys: Phonebook, Guest Service (Hotel/Motel), Trunk Status, Alarm, SMDA, Select Option, or blank (no application).
Attendant Console Status Display	Attendant Console Status Display on each attendant console displays various parameters such as Day/Night Service, Attendant Status, and Alarm Status.
Attendant Directory Number	Attendant Directory Number lets you dial an attendant directory number (typically "0") to reach the attendant. Separate directory numbers can be programmed for each attendant console.
Attendant Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features. SUPERSET 7000 consoles can Hold up to six calls; SUPERCONSOLE 1000 consoles can Hold up to eight calls.
Attendant Identity Information Display	Attendant Identity Information Display lets you view the console's prime directory number, the PB software version, and the console's hold slot number.
Attendant Lockout	Attendant Lockout prevents the attendant from re-entering a call once the attendant has released.
Attendant Messaging	Attendant Messaging lets you activate a message waiting condition on a station from the attendant console. The condition can be queried or canceled by the attendant or by a station user with the appropriate Class of Service.
Attendant Metered Calls	Attendant Metered Calls lets you use the attendant console to track the cost of outgoing trunk calls.
Attendant Position Busy-Out	See Attendant Busy-Out (Console), page 63.
Attendant Recall	Attendant Recall automatically alerts the attendant when a trunk call has been extended to an idle station and not answered within a specified time-out period or when a call on Hold at the console has not been answered within a selected time.
	Page 2 of 14

Standard Features (continued)	
Feature Name	**Description**
Attendant Ringer Control	Attendant Ringer Control lets you mute the attendant console ringer. When the attendant console ringer is muted, incoming calls continue to be indicated by the Call Waiting prompt at the top of the display.
Attendant Serial Call	Attendant Serial Call automatically returns a call to the attendant console when the caller finishes with the called party.
Attendant Setup and Cancellation of Station Features	The attendant can setup and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder.
Attendant Station Busy-Out	See Attendant Busy-Out (Station), page 63.
Attendant System Login	The attendant has access to some PBX programming functions from the attendant console. To access these programming functions, the attendant must log on.
Attendant Tone Signaling	Attendant Tone Signaling lets the attendant send tones over the circuit once a call has been established.
Attendant Trunk Group Busy Status	Attendant Trunk Busy Status Display lets you display and/or print the busy status of the system trunk groups from the attendant console.
Auto-Answer	Auto-Answer lets you automatically answer calls that ring your Prime line.
Auto-Hold	Auto-Hold lets you automatically place an active call on Hold when you press a line key to originate or receive another call.
Automatic Route Selection (ARS)	Automatic Route Selection (ARS) simplifies local and long distance dialing by automatically selecting the most convenient and cost-effective route and by inserting and/or deleting the proper routing digits.
Broadcast Groups	See Groups - Key System and Multicall, page 69.
Broker's Call	Broker's Call lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.
Busy Dial Through	Busy Dial Through lets you dial a feature access code sequence when a busy condition is encountered. See Callback, page 66, and Camp-on, page 67.
Busy Override	See Override, page 72.
Calculator	Calculator lets you use your telephone as a basic four function calculator by using the telephone keypad, display, and softkeys.
Calibrated Flash	See Flash - Calibrated, page 69.
	Page 3 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Callback	Callback lets you request that the system notify you when a busy line becomes idle or when an unanswered station goes off-hook and on-hook.
Callback - System Programmable	Callback - System Programmable lets you program the destination of a matured callback set against a key line or multicall line group.
Call By Name	See Phonebook, page 72.
Call Coverage	Call Coverage is provided through a combination of features: Call Rerouting, Call Forward, Do Not Disturb, and Answer Plus™ - Mitel Call Distribution.
Call Duration Display	Call Duration Display provides you with a display of the call duration for incoming and outgoing calls in one minute increments (starting at 0:00) from the beginning of the call to the end of the call.
Call Forward	Call Forward lets you redirect incoming calls to an alternate number.
Call Forward - Cancel All	Call Forward - Cancel All lets you cancel all types of Call Forward.
Call Forward - Follow Me - End Chaining	Call Forward - Follow Me - End Chaining ensures that calls are not further redirected.
Call Forward - Follow Me - Reroute When Busy	Call Forward - Follow Me - Reroute When Busy forwards the call to the original set's First Alternative Rerouting if the call forward destination is busy.
Call Forward - Forced	Call Forward - Forced lets you manually redirect an incoming call on your Prime or private line to another number.
Call Forward - Override	Call Forward - Override lets you bypass any Call Forward condition that is set at the station that you are calling.
Call Hold	See Hold, page 69.
Call Park	Call Park lets the attendant Hold a call so that a telephone user can remotely retrieve the call.
Call Pickup	Call Pickup lets you answer an incoming call that is ringing at another station.
Call Privacy	Call Privacy protects a call from audible Call Waiting tones, as the result of a camp-on, and prevents intrusion of any kind (for example busy override).
Call Release	See Release, page 73.
Call Rerouting	Call Rerouting lets the system redirect calls to alternate answering points or devices under specified conditions. Call Rerouting may be used to redirect calls always (in Day, Night 1, and/or Night 2 mode) or under busy, no answer, or Do Not Disturb conditions.
	Page 4 of 14

Standard Features (continued)	
Feature Name	**Description**
Call Split	See Conference Split, page 67.
Call Swap	See Swap, page 74.
Call Transfer	See Transfer, page 75.
Call Waiting Swap	Call Waiting Swap lets you use the switch hook to alternate between two calls when a party is in Call Waiting for your station or when you have a call on Consultation Hold.
Camp-on (Call Waiting)	Camp-on, or Call Waiting, lets you notify a busy party that you are waiting. An attendant may also put a call through to a busy station to indicate they are waiting. Upon hearing the Call Waiting tone, the busy party can either respond or finish the current call.
Camp-on Tone Security	Camp-on Tone Security prevents you from hearing Camp-on tone. If any party in a call has this option enabled, no Camp-on tone is returned to anyone in the call.
Class of Restriction	Class of Restriction (COR) limits a station's access to specified numbers. A station may have three CORs (Day/Night1/Night2 service), and the COR may also be changed by using a Verified Account Code.
Class of Service	Class of Service (COS) defines a station or trunk's feature and timer options. A station or trunk may have three COSs (Day/Night1/Night2 service), and the COS may also be changed by using a Verified Account Code.
Clear All Features	Clear All Features lets you cancel most of the features activated on your extension or another user's extension.
Conference	Conference lets you connect three or more people into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.
Conference Split	Conference Split lets you separate a 3-party conference so that you can speak privately with one of the parties. While you are speaking privately with one party, the other party is on Consultation Hold.
Data Applications	Data Applications include many data features, which provide data-switching facilities for local and remote data terminals, and/or computers.
Date and Time	The date and time is set through Windows NT. This data appears on all Station Message Detail Recording (SMDR), traffic measurements, data dumps, SUPERSET display telephones, and attendant consoles.
Day/Night Service Control	See Night Service, page 71.
	Page 5 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Dial Pulse Signaling	The PBX can accept or generate rotary dial pulses. The system automatically detects which type of signaling is used by an individual station when a call is dialed. It will automatically outpulse rotary dial or DTMF signals according to the requirements of the particular interconnection.
Dial Tone	You will normally hear continuous dial tone when you lift the handset. You will hear discriminating dial tone (also called interrupted dial tone) or transfer dial tone under certain conditions.
Dial Tone - Outgoing Calls	The PBX can provide a pseudo-CO dial tone to prevent possible confusion to station users.
Dialed Number Editing	Dialed Number Editing lets you edit numbers during dialing.
Dialing - Conflicting Numbers	The system can differentiate between conflicting numbers such as 1-0-0-0-0 and 1-0-0-0. In this example, if the 5th digit is not dialed within a time-out period, the system assumes that the dialed sequence is complete and makes the call.
Direct-In Lines (DIL)	Direct-In Lines (DIL) allow incoming trunks to be assigned to a specific station or hunt group so that calls from the trunk ring the station or hunt group directly.
Direct Inward Dialing (DID)	Direct Inward Dialing (DID) allows incoming calls on designated trunks to directly access predefined stations (or other answering points) on the PBX.
Direct Inward System Access (DISA)	Direct Inward System Access (DISA) lets external callers access the PBX by using a special trunk. The system sees the DISA trunk as a station with its own Class of Service and Class of Restriction. Calls that enter the system on DISA trunks have access to a variety of system features. In all cases, the DISA trunk can be assigned account codes to provide a high degree of security or additional options.
Direct Outward Dialing (DOD)	Direct Outward Dialing (DOD) lets you make external calls without the assistance of the attendant.
Direct Page	Direct Page allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce, page 71.
Do Not Disturb	Do Not Disturb (DND) lets you place your set in an apparent busy condition without affecting the outgoing functionality. If someone calls your set while DND is activated, he or she will hear special busy tone.
DTMF Keypad Support	DTMF Keypad Support lets ONS/OPS extensions use all 16 keys on a 4x4 DTMF keypad. The additional row of four keys (ABCD) is used to access features in the system.
	Page 6 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
DTMF To Rotary Dial Conversion	See Tone-To-Pulse Conversion, page 75.
Display Contrast Control	Display Contrast Control lets you adjust the contrast of the alphanumeric display.
Feature Keys	Feature Keys let you activate features without dialing feature access codes.
Flash - Calibrated	Calibrated Flash provides an alternative method of generating a Switchhook Flash.
Flash - Switchhook	Switchhook Flash lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features.
Flash - Trunk	Trunk Flash lets you single or double flash a trunk in order to access Centrex™ features.
Flexible Answer Point	Flexible Answer Point lets station and console users program a night answer point for their incoming trunk calls.
Ground Button	A Ground Button (Recall Button) lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features. The Ground Button provides an alternative method of producing a Switchhook Flash.
Group Page	Group Page lets you page a group of telephones over their built-in speakers.
Groups - Key System and Multicall	Key System Groups and Multicall Groups let multiple telephones share the same extension number. Incoming calls ring all of the idle stations, and the stations stop ringing when one member answer the call.
Handset Receiver Volume Control	Handset Receiver Volume Control lets you adjust the volume of the handset receiver.
Handsfree Operation	Handsfree Operation lets you use your telephone without lifting the handset.
Headset Operation	Headset Operation lets you use a Headset to make and receive telephone calls.
Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features. The call can be either retrieved at the originating telephone or another telephone.
Hotline	Hotline limits your access to a designated answer point. The system automatically dials the answer point when you go off-hook. The designated answer point can be another station, an attendant, a trunk, or a hunt group.
	Page 7 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Hunt Groups	Hunt Groups let you dial a pilot number and have the call completed to the first idle station in a group of stations. Any station within a Hunt Group may be accessed directly by dialing the station number.
Intercept Handling	Intercept Handling lets the PBX control what happens to a call when the call cannot be completed to the required destination. A call may be routed to a tone or to a directory number. Two alternate destinations may be programmed for each condition.
Interconnect Restrictions	Each peripheral device is assigned an Interconnect Number that is used to restrict one device from connecting with another. Interconnect Restrictions can be used to restrict access to certain trunks, stations, or equipment (i.e. data communications equipment). The restriction is also a function of the direction of the call.
Key System Groups	See Groups - Key System and Multicall, page 69.
Language Change	Language Change lets you change the language of the telephone softkeys and prompts to any one of the following languages: English, French, Italian, German, Spanish, or Dutch.
Line Types and Appearances	Line appearance keys are single or shared lines that appear on the SUPERSET telephone programmable keys. There are three types of lines: Prime, Non-Prime, and No Where Prime.
Line Appearance Ring Types	Each line appearances can be programmed to ring in a different manner.
Maintenance	The PBX provides extensive maintenance coverage. All types of peripheral hardware are periodically tested by the system. Maintenance users may also test individual circuits on demand.
Meet Me Answer	Meet Me Answer lets a paged party respond to a Group Page even if they do not know the identity or location of the paging party.
Messaging - Advisory	Messaging - Advisory lets you select a short advisory message to show display set users who call your telephone.
Messaging - Callback	Callback Messaging lets you leave a callback message on a telephone when the called party is busy or does not answer. When you receive a callback message, you can review the message on the display (if applicable) and/or call the sender back.
Messaging - Dialed	Dialed Messaging lets you leave a message-waiting indication on a telephone. When you receive a message-waiting indication, you call your message taker to accept the message.
Mixed Station Dialing	Mixed Station Dialing lets you use both rotary dial and DTMF telephones within the system and on the same line.
	Page 8 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Multicall Groups	See Groups - Key System and Multicall, page 69.
Multiple Consoles	See Attendant Consoles (Multiple), page 64.
Music	Music lets you listen to the Music On Hold music source through the speaker of the telephone.
Music On Hold	Music On Hold provides callers with music while they are waiting for a call to be completed. Music On Hold is provided when a call is on Hold, when a call is transferred to a busy party, or when a call is in Call Waiting for a station. The music source is provided by the customer.
NI3 Calling Name Delivery	NI3 Calling Name Delivery allows the called party to see the name of the caller on the telephone display screen if the caller has programmed Calling Name to "Allow" through IMAT. NI3 supports both incoming and outgoing calls for the PBX T1 card and is supported by the Dual T1/E1 card.
Networking	The PBX supports both analog and digital networking. See Node ID Recognition, page 71, and Uniform Numbering Plan, page 75.
Night Service	Night Service lets you redirect calls to alternate answer points for individual trunks. The answer point used depends on the selected mode of operation (Day, Night 1, or Night 2).
Night Service - Automatic	Automatic Night Service places the system into Night service automatically if all attendant consoles are unable to receive calls or if all attendant consoles are inactive and the time-out period has expired.
Node ID Recognition	Node ID Recognition lets a system in a network determine if an incoming call applies to it or to another system in the network.
Non-Busy Station	Non-Busy Station lets you program an extension never to return busy tone. This feature is used for special situations such as emergency stations.
Non-DID Extension	Non-DID Extension allows the system to support sets that are not directly accessible to DID trunks. These calls are transferred to Non-DID Extensions by an Intercept Handling point (such as an attendant or a station).
Off-Hook Voice Announce	Off-Hook Voice Announce lets you receive a Direct Page during a handset or headset call. See Direct Page, page 68.
Overlap Outpulsing	Overlap Outpulsing reduces post-dialing delay when trunk calls are originated. Once a route has been determined by ARS, a trunk is seized and dial pulses or tones are outpulsed to the CO. These outpulses are sent before the user has finished dialing to allow faster call setup on analog trunks.
	Page 9 of 14

Standard Features (continued)	
Feature Name	**Description**
Override	Override lets you enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.
Override Security	Override Security prevents users from using Override on your station.
Paging	Paging lets you connect to loudspeaker/paging equipment to access individual paging zones or all paging zones simultaneously. Before you are connected to the paging equipment, you will hear a two-second burst of tone.
Phonebook	Phonebook lets you locate and telephone a system user based on his or her name, extension number, department, and/or location.
Printer Support	The PBX has complete RS-232 printer flexibility. Any printer port may be programmed for any application. The system supports both system printers for its own applications (such as SMDR and maintenance) and dedicated data communications printers.
Priority Queuing	Priority Queuing ensures that calls are handled in order of priority. When internal or external callers must wait for calls to be completed, they are placed into a queue and assigned an access priority.
Privacy Release	Privacy between users who share line appearances in key systems groups is automatic. The privacy release feature allows users to release privacy during a call to allow another member of the key system group to intrude on the call.
Pulse-To-Tone Conversion	Pulse-To-Tone Conversion automatically converts rotary dial pulses from stations, lines, and trunks to DTMF tones on outgoing trunks that have been programmed as DTMF trunks.
Recall	Recall lets an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out Recalls occur for parties who were transferred to busy stations or who were placed on Hold.
Recall Button	See Ground Button, page 69.
Redial	Redial lets you automatically dial the last number that you manually dialed.
Redial - Saved Number	Redial - Saved Number lets you save a number for future dialing. The number remains saved until a replacement number is saved.
	Page 10 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Release	Release lets you forcibly release from an attempted connection to an external party without going on-hook. Release is useful when you encounter a busy or unavailable external party that you are attempting to add to a Conference.
Reminder	Reminder lets you program your set to ring and provide a message at a specified time within the 24-hour period.
Ringer Control	Ringer Control lets you adjust the volume and pitch of the telephone ringer.
Ringing - Discriminating	Discriminating Ringing lets you distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences).
Ringing - Discriminating (Optional)	Optional Discriminating Ringing lets you change the Discriminating Ringing patterns on ONS/OPS lines so that you hear internal ringing (1 second on and 3 seconds off) for both internal and external calls.
Ringing Line Select	Ringing Line Select lets you answer any ringing line by going off-hook.
Rotary Dial to DTMF Conversion	See Pulse-To-Tone Conversion, page 72.
Speaker Volume Control	Speaker Volume Control lets you adjust the volume of the telephone speaker.
Speed Call Keys	Speed Call Keys let you store and dial frequently-used numbers by using the personal keys on your telephone.
Speed Call - Pause	When the system encounters a Pause while dialing a Speed Call string, the system ceases dialing for the duration of the Pause. When the Pause ends, dialing resumes.
Speed Call - Personal	Personal Speed Calls let you store and dial frequently-used numbers by using access codes and index numbers.
Speed Call - System	System Speed Call lets you dial stored system numbers.
Speed Dial	See Speed Call.
Station Message Detail Recording (SMDR)	There are two separate SMDR applications: external SMDR and internal SMDR. The external SMDR application allows trunk call data for individual stations to be collected for outgoing and incoming trunk calls. Internal SMDR collects data for calls made between stations.
Station Message Detailed Accounting (SMDA)	Station Message Detailed Accounting (SMDA) lets the system accumulate meter pulses (up to an assigned buffer size) that can be read, printed, and cleared from a console. You can collect meter pulses by using either a device (device meter unit accumulation) or an account code (account code meter unit accumulation).
	Page 11 of 14

Standard Features (continued)	
Feature Name	**Description**
Station-To-Station Dialing	Station-To-Station Dialing lets you dial any other station directly.
Suite Service	Suite Service is provided by Pickup Groups. (Set Auto Answer On to YES.)
SUPERSET Loop Test	The SUPERSET Loop Test lets you verify the operation of the telephone keys and displays and the integrity of the data path to the switch. The tests are performed from the set (normally after initial system installation).
Swap	Swap lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.
Switchhook Flash	See Flash - Switchhook, page 69.
System Access Authorization	Administrative access to the PBX is controlled by passwords. Different passwords are assigned for each of the five levels of access.
System Alarm Indications	See Alarms and Attendant Console Status Display, page 64.
System Fail Transfer	See System Fail Transfer.
Tandem Trunking	The PBX can transparently interconnect trunk circuits originating from one CO or PBX and terminating on another (tandem trunking) without attendant intervention.
Telephone Directory - Privacy Option	Any extension number in the system telephone directory can be designated as private. When an extension number is private, the number is not displayed on other users' telephones.
Tie Trunk Support	Tie trunks terminate on the attendant console, at station sets, in hunt groups, or on night bells. They may also be arranged as dial-in tie trunks or tandem trunks. Like CO trunks, tie trunks are arranged in groups.
Timed Reminder	See Reminder, page 73.
Toll Control	Toll control allows or denies access to specified routes, CO exchanges, and directory numbers.
Tone Demonstration	Tone Demonstration lets you hear the tones provided on the PBX.
Tone Detection	The PBX can detect and analyze call progress tones that originate from the central office during the course of a trunk call.
Tone Plan Flexibility	Call progress and supervisory tones generated within the system are programmed to meet the requirements of the telephone authorities of the country in which the PBX is installed.
	Page 12 of 14

Standard Features (continued)	
Feature Name	**Description**
Tone-To-Pulse Conversion	Tone-To-Pulse Conversion automatically converts DTMF tones from stations, lines, and trunks to rotary dial pulses on outgoing trunks that have been programmed as rotary dial trunks.
Transfer	Transfer lets you move a call from one telephone to another. Before completing a Transfer, you can consult privately with the third party and swap between private conversations with each of the parties.
Transmission Tests	Transmission Tests let you perform the following tests on a trunk: milliwatt test, balance test, and 100 test.
Trunk Access	Trunk Access lets you access a specific trunk directly. No toll control or ARS checking is done when you use Trunk Access. This feature is used when a maintenance phone is required.
Trunk Answer From Any Station (TAFAS)	Trunk Answer From Any Station (TAFAS) lets you answer any call that rings a night bell. Once you answer the call, you can use any of the features that are normally available at the station.
Trunk Busy-Out	Trunk Busy-Out lets you busy-out a specific trunk. When you perform a Trunk Busy-Out, the trunk is busied out if it is idle; if the trunk is in use, it is busied out as soon as it becomes idle. When you busy-out the trunk, it cannot be accessed.
Trunk Group Hunting	Trunk Group Hunting lets you search for trunk groups in either a terminal or circular pattern. In terminal hunt group, trunks are always selected in a predetermined order. In a circular hunt group, trunks are selected in a distributed manner (the first free trunk after the last one used becomes the new first choice).
Trunk Labels	Trunk Labels may be assigned to individual trunks or groups of trunks. When a trunk call appears at an attendant console, the trunk label and trunk number are displayed.
Trunk Select - Direct	Direct Trunk Select lets you access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without the need for trunk access codes.
Trunk Support	The PBX supports most public network trunk types (both analog and digital).
Uniform Numbering Plan	The PBX supports the use of a network Uniform Numbering Plan that allows you to use the same digits to reach a station from any location in the network.
Universal Port Orientation	All peripheral interface ports are identical; as a result, the system is flexible and can accommodate various different system configurations.
	Page 13 of 14

STANDARD FEATURES (CONTINUED)	
FEATURE NAME	**DESCRIPTION**
Voice Mail	Most voice processing systems work in conjunction with the PBX. The system provides the following voice processor interfaces: • Voice Mail - COV Interface • Voice Mail - Digital E&M Interface • Voice Mail - E&M Interface • Voice Mail - ONS Interface. The PBX will typically use MSDN or MSAN facilities to network other PBXs. Various types of tie trunks are also used to link PBXs; however, they offer less functionality between sites. These facilities can be configured to provide voice mail functionality from a centralized voice processor. For more information, see the Voice Mail - Centralized E&M Interface section of the Technical Documentation CD-ROM.
	Page 14 of 14

OPTIONAL FEATURES

The following table lists the optional features available on the PBX.

OPTIONAL FEATURES	
FEATURE NAME	**DESCRIPTION**
ACD 2000 Extended Agent Groups	The ACD 2000 feature package lets you program a maximum of 64 agent groups with up to 150 agents in each group. By using the ACD 2000 Extended Agent Groups feature package, you can assign up to 500 agents to each group; however, the maximum number of agent groups is reduced to 32.
ACD 2000 Skill-Based Routing	Each agent in an agent group is assigned a skill level. Calls to the group are routed to the most skilled available agent. If agents of equal skill are available, the call is routed to the longest-idle agent. To facilitate skill-based routing, agent IDs can appear in more than one agent group.
Advanced Analog Networking	Provides calling line identification and travelling class marks across analog trunks.
Advanced ARS	Allows day and time zones, route plans, and ARS assignment to be programmed.
Advanced Data	Enables data transceiver functionality for access to all of the DTRX features.
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.
	Page 1 of 5

Optional Features (continued)	
Feature Name	**Description**
ANSWER PLUS Automatic Attendant	Allows an external PBX environment caller to dial through to an extension without having to go through an attendant.
ANSWER PLUS Automatic Call Distribution I	Provides the following features: - silent monitoring - agent help - agent log on and log out - modified DND for ACD agents - ongest idle agent queuing - real time event records - work timer - programmable threshold alert - RAD groups
ANSWER PLUS Automatic Call Distribution II (ACD 2000)	Consists of four main components: call distribution, agent mobility, management and reporting, feature configuration and administration. Each of these components offers many features not available with ANSWER PLUS - Automatic Call Distribution.
ANSWER PLUS - Mitel Call Distribution	Permits the use of Recorded Announcement Devices (RADs) and a uniform call distribution to hunt groups.
Attendant Language Selection	Enables attendant to choose language of operation for the attendant console (English, French, German, or Italian).
ANI / DNIS	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.
Autovon	Allows the connection of the PBX with Autovon networks (defense switched networks and Canadian switched networks) for outgoing and incoming calls.
CLASS/CLIP Station Side Software Support	Allows ONS CLASS/CLIP sets using CLASS/CLIP protocol to receive Caller Line Identification Delivery (CLID) information, and the time and date of a call.
COV	Allows you to program SUPERSET 3 and SUPERSET 4 telephones as well as voice mail applications that require a COV interface.
DASS II Voice I	Allows basic calls to be made from the PBX to a DASS II protocol Central Office, using CEPT Digital Trunks and DASS II signaling.
DNI	Allows you to program Mitel digital network devices including SUPERSET telephones, attendant consoles, and datasets.
Direct Station Select/Busy Lamp Field (DSS/BLF)	A Busy Lamp Field (BLF) allows the status of a directory number to appear on the line status indicator of a SUPERSET or Programmable Key Module. The monitored device may be on the same PBX or another PBX within the same cluster. The key associated with the busy lamp acts as a Direct Station Selection (DSS) key.
	Page 2 of 5

Optional Features (continued)	
Feature Name	**Description**
Emergency Services	Allows an Emergency Services number to be dialed, which sends a Customer Emergency Services ID (CESID) from the PBX to the Public Safety Answering Point (PSAP). The CESID is used as a key in the Automatic Location Information (ALI) database to retrieve a database record indicating the precise location of the caller.
Feature Level Optioning	Selected features are grouped together under purchasable levels (they cannot be purchased separately). Feature Level 1 - includes Networked Group Page and Hold on Hold.
Flexible Dimensioning	Allocates database memory to each feature resource. The amount of memory determines the maximum size of the feature resource; the system borrows memory from other resources that are not in use.
HCI®/CTI™ Advanced Telephony	Allows monitoring of the activity and state transitions of extensions.
HCI/CTI Basic Telephony	Permits a host computer application to initiate and clear calls on behalf of an extension on the PBX through X.409, X.410, and X.25 protocols.
Hotel/Motel	Provides features commonly used by hotels, motels, hospitals, as well as a Property Management Interface.
MNMS Configuration Management I	Supports telephone directory management within the OPS Manager, page 30, application.
MNMS Configuration Management III	Supports the following OPS Manager, page 30, functionality: automated software upgrades (on LIGHT PBXs in the network), scheduled automatic database backups, and automatic data saves and data restores.
MNMS Database Access	Supports the following OPS Manager, page 30, functionality: network moves, adds, and changes, single network PBX support, and moves, adds, and changes templates.
MNMS/SNMP Fault Management I	Supports alarm management within the OPS Manager, page 30, application, including the following features: viewing network alarms, alarm paging, demand paging, and history reports.
MSAN/APNSS	Provides call set-up capabilities between PBXs connected in an MSAN/APNSS network.
MSDN/DPNSS Data	Provides data calls over digital network links, fast data call setup, and nailed-up data calls through MSDN/DPNSS links.
MSDN/DPNSS Public Network Access	Allows or denies access to the public network, preventing users from bypassing the toll network and ensuring a trunk entering a private network cannot re-enter the public network.
	Page 3 of 5

Optional Features (continued)	
Feature Name	**Description**
MSDN/DPNSS Redirection	Allows a call which is awaiting connection or reconnection to an extension to be redirected by the terminating PBX in a network.
MSDN/DPNSS Voice I	Provides the following features: - 2.048 or 1.544 Mbps digital multiplexed interface - fast call setup for voice - 3-party and multi-party conferences - camp-on (call waiting) - transfer to busy - hold, swap capabilities - supervised/unsupervised transfer - no access to public network via private network - all analog trunks functionality - network voice mail functionality.
MSDN/DPNSS Voice II	Provides the following features: - callback - override across a network
MSDN/DPNSS Voice III	Displays calling party's name, trunk labels, and SUPERSET display telephone status messages across a network. Provides some network voice mail functionality.
MSDN/DPNSS Voice IV	Provides Serial Call, Call Split, Route Optimization, and three-party conferencing across digital links.
MSDN/DPNSS Voice V	Provides Stepback and Network SMDR features to operate on nodes of a digital network. Provides some network voice mail functionality.
MSDN/DPNSS Voice VI	Provides the Portable Directory Number feature for OPS Manager, page 30.
Networked ACD	Networked ACD supports ACD functions over a Mitel Switched Digital Network (MSDN). Agent groups at different locations (on different PBXs) may service calls on the network independently of where the call first entered the network.
Peripheral Node Expansion	Allows users to expand the peripheral node capacity to 384 ports, controlling up to 24 peripheral interface cards.
PRI Card	Describes the purchasable options supported by the Dual T1/E1 card. These options include Min/Max, Automated Min/Max, NFAS (Non-Facilities Associated Signaling), D-channel Backup and Remote LAN Access.
Q.SIG	A protocol that allows you to connect a minimum of two PBXs together to form a virtual private network. Q.SIG supports both incoming and outgoing calls for the March Networks 3200 ICP system T1 and E1 cards.
SMDR - External	Collects data for outgoing and incoming trunk calls.
	Page 4 of 5

Optional Features (continued)	
Feature Name	**Description**
SMDR - Internal	Collects data for calls made between stations within the PBX.
Speak@Ease Softkey Support	Provides quick and easy access to the Speak@Ease voice recognition system.
T1/D4	Provides support for T1 Channel Associated Signaling. A Dual T1 card is required.
TAPI Support	Supports MiTAI and TALK TO TAPI computer telephony interfaces.
Traffic Reporting	Provides traffic reports based on system usage to allow better system resource management.
Trunk Group Busy Status	Enables attendants to query the status of trunk groups from the attendant console.
Voicemail Softkeys	Provides the SUPERSET user with a quick and convenient method to access NuPoint Messenger. Access to the system is provided through context sensitive softkeys presented on the SUPERSET.
Wireless Software Option	Provides wireless Symbol NetVision Phones and Data Phones with PBX functionality.
	Page 5 of 5

Security

Toll Control

Comprehensive toll control is an integral part of the Advanced Automatic Route Selection feature package. It allows the customer to restrict user access to trunk routes and/or specific external directory numbers. The key to toll control is the use of Class of Restriction (COR) values.

All systems with any combination of Direct Inward System Access (DISA), integrated auto attendant, or RAD groups and peripheral interfaced auto attendant or voice mail are susceptible to being "hacked into" by external callers. In addition, internal users could abuse telephone privileges by using external call forward, trunk-to-trunk connection without third party, or 1-800 and 1-900 numbers.

It is very important to assign telephone privileges that relate to toll only to the employees who require it to do their job. In addition, a lobby telephone would be denied toll access unless authorized through an attendant.

Class of Service options, such as Individual Trunk Access, bypass all ARS and COR restrictions. Features carrying a risk of toll abuse include:

- Public Network to Public Network Connection Allowed permitting trunks to be connected together without a third party
- Call Forwarding External Destination feature allowing an extension user to forward calls to external trunks
- 800 numbers, usually free calls, but some central offices can allow the reversal of 800 charges, making it a toll call for your company
- 900 numbers, should be restricted from all users except those who require it for their job function.

Another method of toll control is Station Message Detailed Recording (SMDR) which can be used to track internal users and control their calls. Tracking is a deterrent to toll abuse by internal callers; however, these records may also be used to identify account codes. For more information on suppressing private strings from records, see the Digit Modification Assignment section of the Technical Documentation CD-ROM.

Authorized Maintenance Access

Authorized maintenance access provides protection (restricted access) for various administration commands from unauthorized users.

The six levels of authorization are:

- System
- Installer
- Maint2
- Maint1
- Supervisor
- Attendant.

These levels provide a means of differentiating the capabilities of the various types of users who administer the system.

Each authorization level requires entry of a username and associated password. The usernames for levels System, Installer, Maint2, and Maint1 are SYSTEM, INSTALLER, MAINT2, MAINT1 respectively; each is equipped with a default password upon system start.

The usernames associated with levels SUPERVISOR and ATTENDANT are programmed by using CDE procedures at installation. Each level may have a number of usernames associated with it (e.g., ATTENDANT, MARY SMITH, JANE BROWN). The system assigns a different default password for each username within the SUPERVISOR and ATTENDANT authorization levels.

A user logged on at a particular authorization level may permanently alter the password associated with that username.

Attendant functions are available without the need to log on with Attendant authorization.

All systems with modems connected to the maintenance port should have all levels of passwords and usernames changed from the default value on an irregular schedule. In the Form Access Authorization form there are different levels of access (MAINT1, MAINT2, SUPERVISOR, and ATTENDANT). Special attention should be given to these users and only allow access to the required forms.

Another area of concern is the User Authorization Profile form. The name and authorization level is accessible from anywhere (i.e., maintenance terminal and modems). Therefore, the authorization level should be kept to a minimum to keep "hackers" from accessing certain forms through CDE.

Voice mail systems connected directly to modems should employ a surveillance device. Also, most voice mail systems require a password to gain access; therefore, make sure this password is difficult to guess and is changed frequently. Any user no longer authorized to use the system should have password privileges revoked.

For more information, see the RESET PASSWORD, RESET USERNAME, and CHANGE PASSWORD sections of the Technical Documentation CD-ROM.

NETWORK SECURITY

When FTP and Telnet user names and passwords are transmitted over the network, they are not encrypted. Therefore, these passwords and usernames are vulnerable to systems hackers who use tools to "sniff" passwords off of the network.

Where possible, use firewalls to restrict access to the server. However, you should not assume that firewalls will solve all network security problems since many damaging incidents of computer crime are carried out inside the company.

File Transfer Protocol Security

File Transfer Protocol (FTP) is used to transfer data between the PBX and the OPS Manager server client station. You must assign a Windows NT account and password for FTP services at the March Networks 3200 ICP system server.

Note: The PBX has a factory default FTP account of mnms and password mx2000.

Telnet Security

The Telnet username and password allows a technician to access a maintenance session on the PBX. You should restrict Telnet access into the PBX to subnets or specific hosts. To restrict access, enable Internet Security in the Internet Security Assignment form, then enter the subnet addresses and IP addresses that are allowed to Telnet into the system.

Note: By default, Internet Security is not enabled. You should enable it after initial installation.

Mitel Options Password

New Mitel Options Passwords (MOP) are not required for software stream upgrades from Lightware 30 Release 2.0 to further streams, when no new purchasable software options are enabled on the system. If new software options are purchased, a new MOP is always required.

Note: If you are upgrading to a software stream released previous to Lightware 30 Release 2.0, a new MOP is still required.

Maintenance

Printer Support

The March Networks 3200 ICP system can print NT files to any compatible print device on the Network. There is also a parallel printer interface on the server. In addition, the PBX supports a local, serial printer (LPR1) that can be programmed to print Maintenance Logs, SMDR Logs, ACD Logs, Software Logs, Hotel/Motel Logs, Traffic Reports, or CDE forms. By using the OPS Manager Maintenance Terminal, you can also direct PBX printouts to the virtual maintenance printer LPR3.

Alarms

An alarm is an event that takes place when an anomaly is detected and corrective action is required.

Alarm Classes: There are three classes of alarms: CRITICAL, MAJOR, and MINOR. Alarm threshold levels are programmable.

- **Critical:** A critical alarm is a total loss of service which demands immediate attention. A critical alarm invokes system fail transfer.
- **Major:** A major alarm is a fault which affects service to many users. This usually results in a major degradation in service and needs attention to minimize customer complaints.
- **Minor:** A minor alarm is any fault which does not fall in either of the above two classes. Whenever the system is not 100% operational a minor alarm is raised. This normally requires the attention of a repair person but is not urgent. Examples of a minor alarm include the loss of a single line or trunk circuit or the loss of one circuit switch link.

Alarm Routing: All attendants are alerted when an alarm is raised. To determine the alarm status, log on to the maintenance terminal and enter the ALARMS command. An alarm condition is CLEARED when the fault or condition which caused it is corrected, or the threshold is reprogrammed outside of that which caused the alarm.

Controls and Indicators

Circuit Card Indicators

All Printed Circuit Board (PCB) cards in the system have a series of LED indicators and/or numeric displays mounted on their front panels. For maintenance instructions, see the Circuit Card Indicators section of the Technical Documentation CD-ROM. The PCB card LEDs can be grouped into three categories:

- **Card Status LEDs:** common to all cards
- **Circuit Status Bar LEDs:** found on line, trunk, and DTMF Receiver cards

Specifications

Environment

Note: For the server's environmental specifications, refer to the manufacturer's documentation.

Peripheral Cabinet

Storage Environment	
Condition	**Specification**
Temperature	-40° to 150°F (-40° to 66°C)
Humidity	5-95% Relative Humidity, non-condensing
Vibration (FCC Part 68, Sections 6&7)	0.5 g, 5 to 100 Hz, any orthogonal axis 1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress (FCC Part 68, Sections 6&7)	One 20.3 cm (8 inch) drop, each edge and corner adjacent to the rest face
Horizontal Transportation Impact Stress	One shock pulse applied on each face perpendicular to the direction of motion of the transporting vehicle; the shock pulse is a half-sine acceleration 30 g peak, 20 ms duration

Operational Environment	
Condition	**Specification**
Temperature	32° to 122°F (0° to 50°C)
Humidity	5-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	724 BTUs per hour
Air Flow	150 cubic feet per minute at maximum output of the fans
Radiated Emissions	The system meets Class A limits as outlined in FCC Rules, Part 15, Subpart J
Conducted Emissions	The system meets Class A limits as outlined in FCC Rules, Part 15, Subpart J, and complies with conducted emissions standards as outlined in BS800
Acoustic Emissions	Maximum 50 dBA continuous, 75 dB intermittent (<10% duty cycle)
	Page 1 of 2

OPERATIONAL ENVIRONMENT (CONTINUED)	
CONDITION	**SPECIFICATION**
Static Discharge	Withstands 50 discharges of each polarity through a 10 k resistor connected to a 60 pF capacitor charged to 20 kV, and 20 discharges of each polarity through 500 ohm resistor connected to a 100 pF capacitor charged to 10 kV
Lightning Surge	2.5 kV peak, with a maximum rise time of 2 μs and minimum decay time of 10 μs applied to power lead terminals, and 800 V peak with a maximum rise time of 10 ms and minimum decay time of 560 ms applied to outside plant interface terminals
Induction (Normal)	50 Vrms at 60 Hz, open circuit, longitudinal mode (Tip and Ring to ground)
Power line Faults and Line Crosses (Abnormal)	600 Vrms between Tip and Ring or to ground
Flammability	Minimum oxygen index: 28%, as outlined in ASTM D2863-70 and ASTM D28664-74; meets all safety specifications for product type (CSA, UL, and BT) for use in office environment
Note: Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	
	Page 2 of 2

PERIPHERALS

SUPERSET 4000 AND SUPERSET 400 TELEPHONES PROGRAMMABLE KEY MODULES (PKM48 AND PKM12)		
	TEMPERATURE	**HUMIDITY**
Operating Environment	32° to 122°F (0° to 50°C)	0% to 90% RH, non-condensing
Shipping/Storage Environment	-13° to 158°F (-25° to 70°C)	0% to 90% RH, non-condensing

SUPERCONSOLE 1000		
	TEMPERATURE	**HUMIDITY**
Operating Environment	32° to 86°F (0° to 30°C)	20% to 80% RH, non-condensing
Shipping/Storage Environment	-4° to 140°F (-20° to 60°C)	10% to 70% RH, non-condensing

SUPERSET 7000		
	Temperature	**Humidity**
Operating Environment	32° to 104°F (0° to 40°C)	5% to 95% RH, non-condensing
Shipping/Storage Environment	-40° to 122°F (-40° to 50°C)	5% to 95% RH, non-condensing

Single-line DLM		
	Temperature	**Humidity**
Operating Environment	32° to 104°F (0° to 40°C)	5% to 95% RH, non-condensing

Multi-channel DLM		
	Temperature	**Humidity**
Operating Environment	32° to 104°F (0° to 40°C)	5% to 95% RH, non-condensing
Cabinet clearance/ Ventilation	Unrestricted airflow above and below the cabinet is required. If using a racking system, a minimum of 7.5 cm (3 inches) headroom is needed, with forced ventilation used within the racking equipment.	

Wireless Software Option Peripherals

NetVision Phone

The Symbol NetVision Phone operating conditions and specifications are as follows:

NetVision Phone Environment	
Condition	**Specification**
Drop Specification	3.3 ft./1 meter to concrete surface
Temperature	-300 C to 600 C/ -220 F to 1400 F ambient
Humidity	10% to 95%
Accessories	Dual battery charger, hands-free cradle, leather holster, deluxe carrying case

SPECTRUM24 ACCESS POINTS

The Spectrum24 access point operating conditions and specifications are as follows:

SPECTRUM24 AP ENVIRONMENT	
CONDITION	**SPECIFICATION**
Operating Temperature	-4° to 131°F (-20° to 55°C)
Storage Temperature	-40° to 149°F (-40° to 65°C)
Humidity	10% to 95% non-condensing
Weight	1 lbs (0.454 kg)
Shock	40 G, 11 ms, half sine
Electro-static Discharge	meets CE-Mark
Drop	Withstands up to a 30 inch (76 cm) drop to concrete with possible surface marring.

DIMENSIONS AND WEIGHTS

COMPONENT	HEIGHT	WIDTH	DEPTH	WEIGHT
800 Server	17.25 inches (43.8 cm)	8.0 inches (20.3 cm)	16.0 inches (40.6 cm)	45 lbs (20.5 kg)
1500SR Server	10.5 inches (26.7 cm)	17.5 inches (44.5 cm)	23.5 inches (59.7 cm)	65 lbs (29.5 kg)
1400 Server	6.56 inches (195 mm)	19 inches (492 mm)	18 inches (445 mm)	35 lbs
1400SR Server	6.56 inches (195 mm)	19 inches (492 mm)	18 inches (445 mm)	50 lbs
Peripheral Cabinet	19 inches (48.0 cm)	18 inches (45.8 cm)	16.5 inches (42.0 cm)	95 lbs (43.2 kg)
Peripheral Cabinet II	19 inches (48.0 cm)	18 inches (45.8 cm)	16.5 inches (42.0 cm)	95 lbs (43.2 kg)
SUPERCONSOLE 1000	4 inches (10.2 cm)	15.5 inches (39.4 cm)	9 inches (22.9 cm)	5 lbs (2.27 kg)
Single-line DLM	1.4 inches (3.5 cm)	8.1 inches (20.5 cm)	10.6 inches (27 cm)	2.6 lbs (1.2 kg)

WIRELESS APPLICATIONS SOFTWARE OPTIONS PERIPHERALS

COMPONENT	HEIGHT	WIDTH	DEPTH	WEIGHT
Symbol Access points	1.25 inches	7.75 inches	5.5 inches	1 lbs (0.454 kg)
NetVision Phones	5.10 inches/ 13.0 cm	1.75 inches/ 4.5 cm	0.75 inches/ 1.9 cm	5.5 ounces/ 154 grams
Net Vision Data Phones	6 inches/ 15.24 cm	2 inches/ 5.08 cm	1 inch/ 2.54 cm	8 ounces/ 227 grams

Power

Equipment	Power Requirements
AC Control Cabinet, DSU Cabinet, and Peripheral Cabinet For more information, see the AC Power Converter section of the Technical Documentation CD-ROM.	120 Vac, 6 amps 240 Vac The input power is converted to ±5, ±12, -27 and -48 Vdc, and 80 Vac ringing voltage by the power converter (AC)
DC Control Cabinet, DSU Cabinet, and Peripheral Cabinet For more information, see the DC Power Converter section of the Technical Documentation CD-ROM.	-48 Vdc In a DC powered peripheral or SX-2000 MICRO LIGHT node, the -48 V power is used directly. In a control or DSU node, the input power (AC or DC) is converted to ±5 and ±12 Vdc and output by the PSU.
SUPERCONSOLE 1000	The console is powered from the line feed (-48 V). It has an on-board power supply that converts the input voltage to +5, +15 and -8.0 Volts. The on-board power supply draws 27 mA from input supply, + 10% at -48 Volts. The Input voltage range is from 35 to 60 Vdc.
SUPERSET 7000 PC DNIC Card	+5 Vdc ± 5% -5 Vdc ± 10% +12 Vdc ± 5% -12 Vdc ± 10%
Datasets	Dataset receives power from a plug-in transformer which supplies 9 Vac to a power connector on the back of the dataset. Circuitry in the dataset converts this power to the required DC voltages. The digital telephone voice operation receives its power from the PBX.
IP Telephones	Power is supplied to each IP telephone either with a supplied power adaptor or remotely over the CAT 5 cable via the March Networks™ 3300 In-Line Power Unit

Traffic and Performance

Criteria	Result
Busy Hour Call Completions (BHCC)*	1.5197 per second 5471 per hour
Response Time Specification	Delay to Dial Tone 1 s Dial Tone Cut Off Delay 500 ms Post-Dialing Delay 1.5 s Connecting Delay 400 ms
Data Blocking Possibilities	Software <0.0001 Blocking Probability DTMF, Trunks Provisioning dependent
Note: The BHCC will vary according to individual customer configuration and usage.	

DIGITAL TRUNKING

MEASUREMENT	T1/E1 LINKS	ACD AGENTS	CALL/HR	ERLANG	CCS
Both way traffic per station	2	50	832	35.6	1280
Both way traffic per station	4	100	1924	81.44	2932
Both way traffic per agent	2	1	16.31	0.70	25.1
Both way traffic per agent	4	1	19.05	0.81	29.03

ANALOG TRUNKING

The performance targets for 160 ONS/OPS lines and 12% analog trunking with analog signaling are listed in the following table.

MEASUREMENT	LINES	LS/GS TRUNKS	CALL/HR	ERLANG	CCS
Both way traffic per station	160	32	730	26.69	961
Both way traffic per port	160	32	3.51	0.13	4.46

ATTENDANT CONSOLE REQUIREMENTS

NUMBER OF LINES SUPPORTED	AVERAGE NUMBER OF ATTENDANTS REQUIRED		
	LIGHT TRAFFIC (1.4)	MEDIUM TRAFFIC (2.8)	HEAVY TRAFFIC (5.3)
100 200 300 400	1 1 2 2	1 2 2 3	2 3 3 4
Notes 1. These figures based on originating CCS/line. 2. Theoretically, the PBX supports a maximum of 192 extensions and 120 trunks (CEPT channels).			

SUPERSET 7000 Attendant Console Performance

For optimum performance of the SUPERSET 7000 attendant console, observe the following configuration guidelines for each DNI line card:

- Limit traffic to less than 500 calls per hour
- Add a maximum of eight additional display sets per line
- Add up to four non-display sets
- Configure no data or other type of console.

Tone Plan Support

The system supports tone plans for the following countries:

- North America
- United Kingdom.

For more information, see the Tone Plans section of the Technical Documentation CD-ROM.

Capacity Levels, Software Configurations, and Dimensions

The dimensions and feature allocations determine the “size” of the system in terms of its ability to support peripheral devices and features. The capacity levels, software configurations, and dimensions are divided into the following classes:

- MITEL Feature Resource Dimensions, page 95, (MFRDs)
- MSDN/DPNSS Network Resource Dimensions, page 102, (MNRDs)
- MITEL Traffic Capacity Levels, page 103, (MTCLs)
- Application Capacity Levels
  - MITEL HCI Capacity Level, page 104, (MHCL)
  - MITEL TAPI Service Provider, page 104, (MTSPs)
  - MITEL ACD Agent Capacity Level, page 104, (MACLs)
  - MITEL Agent ID Appearance Capacity Level, page 105, (MAIACLs)
  - MITEL Maximum Line appearances Capacity Level, page 105, (MLCLs)
- MITEL Software Applications, page 105, (MSAs).

Options can be purchased individually or as part of a MITEL Software Bundle, page 107, (MSB) or MITEL Core Package, page 107, (MCP).

MITEL Feature Resource Dimensions (MFRDs)

MITEL Feature Resource Dimensions (MFRDs)	
MFRD-A-14	Small Business Package (352 ONS/OPS)
MFRD-A-16	Medium Business Package (1360 ONS/OPS)
MFRD-A-18	General Business Package (2500 ONS/OPS)
MFRD-A-40	32-Station Package
MFRD-A-42	48-Station Package
MFRD-A-44	96-Station Package

The following tables provide the maximum number of resources available for various parameters of each MITEL feature resource dimension (MFRD).

Note: Some system dimensions can be tailored to meet specific business needs by using the Flexible Dimensioning optional feature package.

HARDWARE DIMENSIONS

HARDWARE DIMENSIONS						
PARAMETER NAME	MFRD LEVEL					
	14	16	18	40	42	44
Total Attendant Consoles	8	16	24	2	4	6
DNI Channels (Note 1)	448	864	2368	64	96	192
Data Transceiver Circuits	24	24	48	24	24	24
Modems	2	20	40	4	4	2
Programmable Key Modules (PKMs)	15	30	75	16	24	32
System Ports (Note 2):						
- Datasets	16	16	16	16	16	16
- DTMF Receivers	64	128	128	64	64	64
- Multiline Telephones	144	300	756	32	48	96
- ONS/OPS Lines	352	1360	2504	32	48	96
- Trunks (Note 3)	144	312	628	128	128	128
Tone Detector Circuits	32	32	32	32	32	32

Notes:

1. A DNI line card provides 16 DNI circuits with 2 DNI channels per circuit. Given the total number of DNI channels, you can determine the maximum number of DNI line cards. For example, with MFRD 14, you can have 448 DNI channels or 14 DNI line cards (448 divided by 2x16).
2. Although the system dimensions permit support for these figures, each peripheral cabinet has a physical maximum of 192 ports and each DSU cabinet has a physical maximum of 240 (multiplexed) ports.
3. When you program a trunk card in the System Configuration form, the maximum number of trunks is allocated automatically. For example, for the LS/GS card, 8 trunks will be allocated; for the E&M card, 4 trunks will be allocated. See also Digital links in the Feature Dimensions table.

FEATURE DIMENSIONS

FEATURE DIMENSIONS						
FEATURE NAME	14	16	18	40	42	44
Attendant Console Groups	12	12	24	12	12	12
Attendant Console Calls Waiting	26	26	48	99	99	99
Busy Lamp Monitored Devices	71	135	184	28	42	59
Call Rerouting Always	20	64	176	32	32	32
Call Rerouting 1st Alternates	500	500	500	32	32	32

Page 1 of 3

FEATURE DIMENSIONS (CONTINUED)

FEATURE NAME	14	16	18	40	42	44
Call Rerouting 2nd Alternates	16	16	32	32	32	32
Class of Restriction (COR)	64	64	96	96	96	96
Class Of Service (COS)	64	64	96	96	96	96
Dataset Groups	2	2	2	4	4	4
Default Account Codes	50	100	225	255	255	255
Departments (in Tel Dir)	50	50	700	700	700	700
Digit Modification Tables	256	256	256	256	256	256
Digit Blocks	600	600	2000	2000	2000	2000
Digital Links (Note 1)	8	13	27	2	2	4
DTS Service Numbers	10	32	64	16	24	36
Group Page	4	12	16	2	3	4
Groups - Key System and Multicall	360	600	750	240	360	480
Hunt Groups (see Note 2)	20	64	176	16	24	48
Independent Account Codes	400	572	3000	1000	1000	1000
Locations (in Tel Dir)	30	50	70	70	70	70
Modem Groups	2	10	15	4	4	4
Modems per Modem Group	8	20	40	4	4	4
Paging Zones	2	16	16	4	4	8
Personal Speed Call Users	100	500	500	16	24	48
Personal Speed Call Numbers (avg. 12 digits)	500	2500	2500	420	420	420
Pickup Groups	48	100	200	16	24	48
Routes	200	200	200	200	200	200
Route Lists	128	128	128	128	128	128
Suite Services - Single	124	415	815	16	24	48
Suite Services - Linked	41	138	271	5	8	16
SUPERSET Callback Messages per System (see Note 3)	96	200	500	64	96	144
System Account Codes	16	16	24	16	16	16
System Digit Strings (Note 4)	1049	3306	3306	3306	3306	3306
System Speed Call	150	500	600	600	600	600
Telephone Directory Entries	500	1360	3600	3600	3600	3600
Trunk Groups	20	64	112	64	64	64
Trunks per Trunk Group	60	96	175	64	64	64
Trunk Service Numbers	40	64	150	64	64	64

Page 2 of 3

FEATURE DIMENSIONS (CONTINUED)						
FEATURE NAME	**14**	**16**	**18**	**40**	**42**	**44**
Verified Account Codes	400	400	512	1000	1000	1000

Notes:

1. When programming a digital trunk card, you must not exceed the number of trunks allowed and available within your MFRD level. The trunks are allocated as soon as the digital trunk card is added to the System Configuration form. For example, if you add a Dual T1 (DS1 Formatter) card, 48 trunks are allocated; if you add a Dual E1 (CEPT Formatter) card, 60 trunks are allocated. The Dimension and Feature Display form will show you the number of allocated trunks.
2. With ACD 2000 Hunt Mode, up to 150 telephones may be programmed in one hunt group (at MFRD 12 or higher). A maximum of 64 of the available hunt groups can be programmed as ACD groups. For example, MFRD 24 allows 175 hunt groups, but only 64 of the available 175 hunt groups can be programmed as ACD groups.
3. The number of messages includes both callback messages and voice mail messages.
4. Each digit string pool entry can hold up to seven digits but can be used for only one SWID. Numbers larger than seven digits require more than one digit string pool entry (one for each seven digits).

Page 3 of 3

OTHER PARAMETERS

The parameters described below are also features that are provided with each MFRD package; however, the resource limitations for these features are not MFRD dependant (i.e. the maximum resource values remain consistent across all MFRDs).

Consoles Per Attendant Console Group - The maximum number of attendants in each attendant console group is 15.

Sets per Broadcast Group - The maximum number of telephones per broadcast group is 16, 20, 24, 28, or 32 (depending on the purchased option).

Call Progress Tone Plans - Call progress and supervisory tones generated within the PBX are programmed to meet the requirements of the telephone authority of the country in which the system is installed. The tones generated are not programmed by the installer; they are part of the country option of the core package software load. The system provides 100 different tone plans.

Conferees per Conference - The maximum number of conferees per conference is 5 (any combination of internal or external conferees as long as one is internal).

Conferees Per System - The maximum number of conferees per system is 24.

Conferences Per System - The maximum number of three-party conferences per system is 5.

Dataset Circuit Descriptors - Dataset circuit descriptors are used to define dataset circuits by specifying the parameters used by the various datasets supported by the system. One circuit descriptor is assigned to each similarly configured groups of dataset circuits. Default dataset descriptors for specified device classes and usage types may be used, or individual parameters may be programmed. A total of 32 dataset circuit descriptors may be programmed.

Datasets Per Dataset Group - The maximum number of datasets that can be programmed into one dataset hunt group is 50.

Day Zones per Week - There are 3 day zones available, and each of the day zones can be divided into 4 time zones (12 individual day/time zones). These zones are used with the Automatic Route Selection feature (ARS) and the Advanced Automatic Route Selection feature package (MSA 2) to implement Day and Time Zones.

DTE Terminal Profiles - DTE (data terminal equipment) terminal profiles are used with the advanced data application. There are 16 DTE terminal profiles available.

Sets per Hunt Group - The maximum number of telephones allowed in each hunt group is 64; however, with ACD 2000 hunt mode at MFRD 12 or higher, up to 150 telephones may be programmed in one hunt group.

Intercept Numbers - Intercept numbers (maximum of 32) are used to control what happens to a call when the call cannot be completed to the required destinations. A call may be routed to a tone or to a directory number. Two alternate destinations may be programmed for each condition.

Interconnect Numbers - Each peripheral device is assigned an interconnect number (maximum of 64) which is used to bar the connection of one device and another. The interconnect number is an index to the Interconnect Restriction Table that is programmed in CDE.

Multiline Set Status Message Languages - Advisory (status) messages can be programmed in different languages on multiline display telephones. The languages supported are English, French, Italian, German, Spanish, or Dutch. Only three languages can exist on the system at one time.

Multiline Set Status Messages per Language - Up to 20 advisory (status) messages in each of three languages can be programmed. Each advisory message may be up to 13 characters in length.

Node Identifiers - The node identifiers let each PBX operate as a tandem switch in a network to determine if an incoming call applies to it or to another system in the network. Up to 5 local node identifiers and 50 remote leading digits are permitted. Each node identifier is a number with 7 or fewer digits.

Sets per Pickup Group - A maximum of 75 users can be programmed in each pickup group.

Routes per Route List - There can be up to 6 routes in each route list.

Route Plans - Route plans provide a method of presenting different routes to calls as a function of day and time. There are 32 route plans.For additional information, see the Route Plan Assignment form.

Station Circuit Descriptors - Station circuit descriptors are used to assign the operational (signaling and timing) parameters to ONS and OPS circuits. They are not used for DNI circuits. There is a maximum of 16 station circuit descriptors available.

TAPI - The system supports a maximum of 100 TAPI telephones per peripheral cabinet to a maximum of 200 TAPI telephones per main control (both limited to an average of 10 calls/hour/user).

SUPPORTED MFRDS (NO LONGER SOLD)

The following MFRDs are no longer sold; however, they are supported:

- MFRD-A-02
- MFRD-A-03
- MFRD-A-07.

HARDWARE DIMENSIONS

PARAMETER NAME	MFRD LEVEL		
	2	3	7
Attendant Consoles	8	8	16
DNI Channels (Note 1)	448	704	864
DTRX Service (Data Transceiver Circuits)	24	24	24
Modems	8	12	20
Programmable Key Modules (PKMs)	15	25	30
System Ports (Note 2)			
- Datasets	64	96	112
- DTMF Receivers	64	88	128
- Multiline Sets	144	240	300
- Single Line Sets (ONS/OPS Lines)	352	752	1360
- Trunks (Note 3)	144	204	312
Tone Detector Circuits	32	32	32

Notes:
1. A DNI line card provides 16 DNI circuits with 2 DNI channels per circuit. Given the total number of DNI channels, you can determine the maximum number of DNI line cards. For example, with MFRD 2, you can have 448 DNI channels or 14 DNI line cards (448 divided by 2x16).
2. Although the system dimensions permit support for these figures, each peripheral cabinet has a physical maximum of 192 ports and each DSU cabinet has a physical maximum of 240 (multiplexed) ports.
3. When you program a trunk card in the System Configuration form, the maximum number of trunks is allocated automatically. For example, for the LS/GS card, 8 trunks will be allocated; for the E&M card, 4 trunks will be allocated. See also Digital links in the Feature Dimensions table.ports.

FEATURE DIMENSIONS

FEATURE NAME	2	3	7
Attendant Console Groups	12	12	24
Attendant Console Calls Waiting	26	26	48
Busy Lamp Groups (Monitored Devices)	71	135	184

Page 1 of 3

FEATURE DIMENSIONS (CONTINUED)			
FEATURE NAME	**2**	**3**	**7**
Call Reroute Always	20	36	64
Call Reroute 1st Alternates	60	110	256
Call Reroute 2nd Alternates	16	16	16
Class of Restriction (COR)	64	64	64
Class Of Service (COS)	64	64	64
Dataset Groups	10	10	10
Default Account Codes	50	70	100
Departments (in Tel Dir)	50	50	50
Digit Modification Tables	256	256	256
Digit Blocks	600	600	600
Digital Links (Note 1)	6	8	10
DTS Service Numbers	10	16	32
Group Page groups	4	8	12
Hunt Groups (Note 2)	20	36	64
Independent Account Codes	400	400	512
Locations (in Tel Dir)	30	35	50
Modem Groups	4	6	10
Modems per Modem Group	8	12	20
Paging Groups (Zones)	2	2	16
Personal Speed Call Users (with 10 Speed Calls per user)	100	200	500
Pickup Groups	50	70	100
Routes	200	200	200
Route Lists	128	128	128
Speed Call Digit String (avg. 12 digits)	500	1000	2500
SUPERSET Callback Messages per System (see Note 3)	96	160	200
System Account Codes	16	16	16
System Digit Strings (Note 4)	1049	2027	3306
System Speed Call	150	250	500
Telephone Directory Entries	500	900	1360
Trunk Groups	20	34	64
Trunks per Trunk Group	60	70	96
Trunk Service Numbers	40	50	64
			Page 2 of 3

Feature Dimensions (continued)			
Feature Name	2	3	7
Notes: 1. When programming a digital trunk card, you must not exceed the number of trunks allowed and available within your MFRD level. The trunks are allocated as soon as the digital trunk card is added to the System Configuration form. For example, if you add a Dual T1 (DS1 Formatter) card, 48 trunks are allocated; if you add a Dual E1 (CEPT Formatter) card, 60 trunks are allocated. The Dimension and Feature Display form will show you the number of allocated trunks. 2. With ACD 2000 Hunt Mode, up to 150 telephones may be programmed in one hunt group (at MFRD 12 or higher). A maximum of 64 of the available hunt groups can be programmed as ACD groups. For example, MFRD 24 allows 175 hunt groups, but only 64 of the available 175 hunt groups can be programmed as ACD groups. 3. The number of messages includes both callback messages and voice mail messages. 4. Each digit string pool entry can hold up to seven digits but can be used for only one SWID. Numbers larger than seven digits require more than one digit string pool entry (one for each seven digits).			
Page 3 of 3			

MSDN/DPNSS Network Resource Dimensions (MNRDs)

MSDN/DPNSS Network Resource Dimensions (MNRDs)		Prerequisites
MNRD-A-02	2 PBXs	MSA-A-41
MNRD-A-03	3 PBXs	MNRD-A-02
MNRD-A-04	4 PBXs	MNRD-A-03
MNRD-A-05	5 PBXs	MNRD-A-04
MNRD-A-10	6-10 PBXs	MNRD-A-05
MNRD-A-30	11-30 PBXs	MNRD-A-10
MNRD-A-60	31-60 PBXs	MNRD-A-30

Resource	MSDN/DPNSS Network Resource Dimensions (MNRDs)						
	2	3	4	5	10	30	60
PBXs in the cluster (Note 1)	2	3	4	5	10	30	60
Telephone Directory Records	3000	4500	6000	7500	12000	20000	30000
Remote Device Records	1500	3000	4500	6000	10500	18500	28500
System Dspool Records	3000	4500	6000	7500	12000	20000	30000
Additional Digit Tree Records (Note 2)	167	333	500	677	1167	2055	3167
Locations	150	200	250	250	250	250	250
Departments	700	750	800	850	1200	2000	2000

Resource	MSDN/DPNSS Network Resource Dimensions (MNRDs)
Notes: 1. If the existing resource dimension for the cluster element is larger than the MSDN/DPNSS resource dimension listed, the existing dimension is allocated. 2. The total number of digit tree records is the number of records allocated by the existing resource dimension plus the number of records for the specified MSDN/DPNSS resource dimension.	

MITEL Traffic Capacity Levels (MTCLs)

MITEL Traffic Capacity Level (MTCL)	
MTCL-A-5	100 simultaneous two-party connections
MTCL-A-10	180 simultaneous two-party connections
MTCL-A-15	230 simultaneous two-party connections

Resource	MITEL Traffic Capacity Levels (MTCL)			
	1	**5**	**10**	**15**
Call Processes (Note 1)	120	200	360	460
Callbacks per System	30	40	100	128
Device Campons per System	16	24	48	64
Group Campons per System	5	8	20	30
Hard Holds per System	30	40	100	128
Simultaneous two-party connections	60	100	180	230
Telephone Mode Processes (Note 2)	6	6	12	14
Wake-up Calls in 1 Minute	5	7	13	17
Wake-up Calls in 5 Minutes	21	27	51	67

Notes:
1. A call process is equivalent to one party in a call. For example, in a call where two parties are talking and a third is on hold (consultation call), three call processes are involved; an eight-party conference consists of eight call processes.
2. The following telephone keys (on SUPERSET 4025, SUPERSET 4125, SUPERSET 4150, SUPERSET 4015IP, SUPERSET 4025IP, March Networks 5010 IP and March Networks 5020 IP telephones) are considered Mode Processes: SUPERKEY, Messaging softkey, Phonebook softkey, and Account Code softkey.

Application Capacity Levels

The Application Capacity Levels determine the system's ability to support various software applications and features.

MHCL: MITEL HCI CAPACITY LEVEL

MITEL HCI CAPACITY LEVEL (MHCL)		PREREQUISITES
MHCL-A-1	1 HCI session and 100 call monitors	MSA-A-25 and MSA-A-26
MHCL-A-2	2 HCI sessions and 200 call monitors	MHCL-A-1
MHCL-A-3	3 HCI sessions and 300 call monitors	MHCL-A-2
MHCL-A-4	4 HCI sessions and 400 call monitors	MHCL-A-3
MHCL-A-5	5 HCI sessions and 500 call monitors	MHCL-A-4

MTSP: MITEL TAPI SERVICE PROVIDER

MITEL TAPI SERVICE PROVIDER		PREREQUISITES
MTSP-A-10	1 to 10 users	MSA-A-47
MTSP-A-20	11 to 20 users	MTSP-A-10
MTSP-A-40	21 to 40 users	MTSP-A-20
MTSP-A-60	41 to 60 users	MTSP-A-40
MTSP-A-80	61 to 80 users	MTSP-A-60

MACL: MITEL ACD AGENT CAPACITY LEVEL

MITEL ACD AGENT CAPACITY LEVEL (MACL)		PREREQUISITES
MACL-A-005	1 - 5 ACD agents	MSA-A-40
MACL-A-010	6 - 10 ACD agents	MACL-A-005
MACL-A-015	11 - 15 ACD agents	MACL-A-010
MACL-A-020	16 - 20 ACD agents	MACL-A-015
MACL-A-025	21 - 25 ACD agents	MACL-A-020
MACL-A-030	26 - 30 ACD agents	MACL-A-025
MACL-A-035	31-35 ACD agents	MACL-A-030
MACL-A-040	36-40 ACD agents	MACL-A-035
MACL-A-045	41-45 ACD agents	MACL-A-040
MACL-A-050	46-50 ACD agents	MACL-A-045
MACL-A-060	51-60 ACD agents	MACL-A-050
MACL-A-070	61-70 ACD agents	MACL-A-060
MACL-A-080	71-80 ACD agents	MACL-A-070
MACL-A-090	81-90 ACD agents	MACL-A-080
MACL-A-100	91-100 ACD agents	MACL-A-090
MACL-A-150	101-150 ACD agents	MACL-A-100
Note: If you program agents as members of more than one ACD group the ACD agent resource level may determine the maximum number of agents that you can have. Each agent uses one resource. Each time an agent appears in another group it uses up another resource. For example, one agent in one ACD group uses one resource; one agent in two groups takes up two resources; one agent in three groups takes up three resources, and so forth.		

MAIACL: MITEL Agent ID Appearance Capacity Level

MITEL Agent ID Appearance Capacity Level (MAIACL)		Prerequisites
MAIACL-A-02	2 agent appearances	MSA-A-40
MAIACL-A-04	4 agent appearances	MSB-A-07
MAIACL-A-08	8 agent appearances	MAIACL-A-04

MLCL: MITEL Maximum Line Appearances Capacity Level

MITEL maximum Line appearances Capacity Level (MLCL)	
MLCL-A-16	16 maximum line appearances
MLCL-A-20	20 maximum line appearances
MLCL-A-24	24 maximum line appearances
MLCL-A-28	28 maximum line appearances
MLCL-A-32	32 maximum line appearances

MITEL Software Applications (Optional Features)

MITEL Software Applications (MSAs) are optional feature packages that enhance system functionality. The following MSAs are currently available:

MITEL Software Applications (MSAs)			
Number	Option Name	Prerequisites	Notes
MSA-A-01	COV		
MSA-A-02	Advanced ARS		
MSA-A-03	SMDR - External		
MSA-A-04	Advanced Data		
MSA-A-06	T1/D4		
MSA-A-07	Hotel/Motel		
MSA-A-08	Traffic Reporting		
MSA-A-09	Trunk Group Busy Status		
MSA-A-10	Attendant Language Selection		
MSA-A-11	Advanced Analog Networking		
MSA-A-14	DNI		
MSA-A-15	MSDN/DPNSS Voice I		
MSA-A-16	MSDN/DPNSS Voice II	MSA-A-15 or MSA-A-31	
MSA-A-17	MSDN/DPNSS Voice III	MSA-A-15 or MSA-A-31	
MSA-A-18	MSDN/DPNSS Data	MSA-A-15 or MSA-A-31	
			Page 1 of 3

MITEL SOFTWARE APPLICATIONS (MSAS) (CONTINUED)			
NUMBER	**OPTION NAME**	**PREREQUISITES**	**NOTES**
MSA-A-19	MSDN/DPNSS Public Network Access	MSA-A-15 or MSA-A-31	
MSA-A-20	ANSWER PLUS - MITEL Call Distribution (MCD)		
MSA-A-21	MSDN/DPNSS Voice IV	MSA-A-15 or MSA-A-31	
MSA-A-22	MSDN/DPNSS Redirection		
MSA-A-23	ANSWER PLUS - Auto-Attendant	MSA-A-20	
MSA-A-24	DASS II Voice I		EMEAAP only
MSA-A-25	HCI/CTI Basic Telephony		
MSA-A-26	HCI/CTI Advanced Telephony	MSA-A-25	
MSA-A-27	ANSWER PLUS - Automatic Call Distribution (ACD)		
MSA-A-31	MSAN/APNSS		
MSA-A-32	MSDN/DPNSS Voice V	MSA-A-15 or MSA-A-31	
MSA-A-33	Autovon		
MSA-A-34	SMDR - Internal		
MSA-A-35	Flexible Dimensioning	MFRD-A-07 (minimum)	
MSA-A-36	ANI/DNIS		
MSA-A-37	MNMS Fault Management I		for Alarms Mgmt.
MSA-A-38	MNMS Configuration Management I		for Directory Mgmt.
MSA-A-40	ANSWER PLUS - Automatic Call Distribution II (ACD 2000)	MSA-A-20	
MSA-A-41	MSDN/DPNSS Voice VI	MSA-A-15	for OPS Manager Portable Directory Number Operation
MSA-A-42	MNMS Configuration Management III		
MSA-A-43	Networked ACD	MSA-A-41	
MSA-A-44	MNMS Database Access		
MSA-A-45	DSS/BLF	MSA-A-41 (for network version only)	Network DSS/BLF must be used with OPS Manager
			Page 2 of 3

MITEL Software Applications (MSAs) (continued)			
Number	**Option Name**	**Prerequisites**	**Notes**
MSA-A-46	ACD 2000 Skill-Based Routing	MSB-A-07	
MSA-A-47	TAPI Support		
MSA-A-48	ACD 2000 Extended Agent Group		
MSA-A-49	Emergency Services	Dual T1/E1 card	
MSA-A-50	Peripheral Node Expansion		Required when ordering the first Peripheral Expansion Node II in the system
MSA-A-52	Q.SIG	Dual T1/E1 card	
MSA-A-53	Voicemail Softkeys		
MSA-A-62	Speak@Ease		
9125-501-001-NA	PRI Card - Min/Max		
9125-501-002-NA	PRI Card - Auto Min/Max		
9125-501-003-NA	PRI Card - NFAS		
9125-501-004-NA	PRI Card - D-Channel Backup		
9125-501-005-NA	PRI Card - Remote LAN Access		
54000424	Wireless Software Option	3200 Release 2.3	This option and the 3800 Wireless Applications Gateway system are mutually exclusive.
54000094	Suite Services	MSA-A-07	
54000280	Feature Level 1		Includes: Networked Group Page and Hold on Hold
			Page 3 of 3

MITEL Software Bundles (MSBs)

MITEL Software Bundles (MSBs) are groups of MSAs that are suited to specific markets and applications. Depending on your country and region, these packages vary. See your authorized MITEL representative for further details.

MITEL Core Packages (MCPs)

MITEL Core Packages (MCPs) determine the basic configuration of the system. Depending on your country and region, these packages vary. See your authorized MITEL representative for further details.

Reference for Product Name Changes			
Old Name	**New Names**	**Abbreviations**	
Ipera 1000	March Networks™ 3100 Integrated Communications Platform (ICP)	March Networks 3100 ICP	3100 ICP
	March Networks™ 3100 Controller	March Networks 3100 Controller	3100 Controller
	March Networks™ 3100 Expansion unit	March Networks 3100 Expansion unit	3100 Expansion Unit
Ipera 2000	March Networks™ 3200 Integrated Communications Platform (ICP)	March Networks 3200 ICP	3200 ICP
Ipera 3000	March Networks™ 3300 Integrated Communications Platform (ICP)	March Networks 3300 ICP	3300 ICP
	March Networks™3300 Software Rel xx	March Networks 3300 Software Rel xx	Software
	March Networks™ 3300 Controller	March Networks 3300 Controller	3300 Controller
Network Services Unit 3020	March Networks™ 3300 Universal Network Services Unit	March Networks 3300 Universal NSU	3300 Universal NSU
Network Services Unit 3021	March Networks™ 3300 R2 Network Services Unit	March Networks 3300 R2 NSU	3300 R2 NSU
Network Services Unit 3022	March Networks™ 3300 BRI Network Services Unit	March Networks 3300 BRI NSU	3300 BRI NSU
Analog Services Unit 3030	March Networks™ 3300 Universal Analog Services Unit	March Networks 3300 Universal ASU	3300 Universal ASU
Analog Services Unit 3031	March Networks™ 3300 Analog Services Unit	March Networks 3300 ASU	3300 ASU
	March Networks™ 3300 In-Line Power Unit	March Networks 3300 In-Line Power Unit	3300 In-Line Power Unit
QUICK Installation Tool	March Networks™ 3300 Configuration Tool	March Networks 3300 Configuration Tool	3300 Configuration Tool
38XX Gateways			
Ipera Applications Gateway	March Networks™ 3800 Applications Gateway	March Networks 3800 Applications GW	3800 Applications GW
Ipera 2000 with Ericsson Mobile Advantage	March Networks™ 3800 Ericsson Mobile Advantage Gateway	March Networks 3800 Ericsson Mobile Advantage GW	3800 Ericsson Mobile Advan-tage GW
Xipnet	March Networks™ 3800 IP Trunking Gateway	March Networks 3800 IP GW	3800 IP GW
(no old name)	March Networks™ 3800 Ericsson Wireless Assistant Gateway	March Networks 3800 Ericsson Wireless Assistant GW	3800 Ericsson Wireless Assis-tant GW

4XXX Hardware Peripherals			
Networks IA²D	March Networks™ 4500 Integrated Access Device	March Networks 4500 IAD	4500 IAD
5XXX User Interfaces Devices			
Superset 501 IP	March Networks™ 5001 IP Phone	March Networks 5001 IP Phone	5001 IP Phone
Superset 505 IP	March Networks™ 5005 IP Phone	March Networks 5005 IP Phone	5005 IP Phone
Superset 510 IP (4015 IP)	March Networks™ 5010 IP Phone	March Networks 5010 IP Phone	5010 IP Phone
Superset 520 IP (4025 IP)	March Networks™ 5020 IP Phone	March Networks 5020 IP Phone	5020 IP Phone
Webset	March Networks™ 5140 IP Appliance	March Networks 5140 IP Appliance	5140 IP Appliance
Small Office Conf Unit	March Networks™ 5305 IP Office Conference Unit	March Networks 5305 IP Office Conference Unit	5305 IP Conference Unit
Board Room Unit	March Networks™ 5310 IP Board Room Conference Unit	March Networks 5310 IP Board Room Conference Unit	5310 IP Conference Unit
PC Console (SC 2000)	March Networks™ 5550 IP Console	March Networks 5550 IP Console	5550 IP Console
PKM12	March Networks™ 5410 Programmable Key Module	March Networks 5410 PKM	5410 PKM
PKM48	March Networks™ 5415 Programmable Key Module	March Networks 5415 PKM	5415 PKM
SIM1	March Networks™ 5421 Interface Module	March Networks 5421 IM	5421 IM
SIM2/AIM	March Networks™ 5422 Analogue Interface Module	March Networks 5422 AIM	5422 AIM
IrDA Module	March Networks™ 5423 IrDA Module	March Networks 5423 IrDA	5423 IrDA
PDA Software	March Networks™ 5810 PDA Application	March Networks 5810 PDA	5810 PDA
Impresa Personal Assistant	March Networks™ 5820 Desktop Assistant	March Networks 5820 Desktop Assistant	5820 Desktop Assistant
SC1000 for IP	March Networks™ 5500 IP Console	March Networks 5500 IP Console	5500 IP Console
SIP Phone (POTS)	March Networks™ 5051 SIP Phone	March Networks 5051 SIP Phone	5051 SIP Phone
SIP Phone	March Networks™ 5055 SIP Phone	March Networks 5055 SIP Phone	5055 SIP Phone

6XXX Applications eComm/eBusiness			
	March Networks™ 6000 Small Business Applications Server (SBAS)	March Networks 6000 Small Business Applications Server	6000 SBAS
Customer Interaction Suite	March Networks™ 6100 Contact Center Solutions (CCS)	March Networks 6100 Contact Center Solutions	6100 CCS
Impresa Workforce	March Networks™ 6120 Contact Center Scheduling	March Networks 6120 Contact Center Scheduling	6120 Scheduling
Impresa iQueue	March Networks™ 6160 Intelligent Queue (IQ)	March Networks 6160 Intelligent Queue	6160 IQ
Impresa Cyber@ED	March Networks™ 6150 Multimedia Contact Center (MCC)	March Networks 6150 Multimedia Contact Center	6150 MCC
Impresa CyberACD	March Networks™ 6110 Contact Center Management (CCM)	March Networks 6110 Contact Center Management	6110 CCM
CyberACD Interactive	March Networks™ 6115 Interactive Contact Center	March Networks 6115 Interactive Contact Center	6115 Interactive
Nurse Dispatch	March Networks™ 6451 Intelligent Dispatch (ID)	March Networks 6451 Intelligent Dispatch	6451 ID
Speak@Ease	March Networks™ 6500 Speech Enabled Applications	March Networks 6500 Speech Enabled Applications	6500 SE Applications
Speak@Ease Atten-dant	March Networks™ 6500 Speech Enabled Attendant	March Networks 6500 Speech Enabled Attendant	6500 SE Attendant
Speak@Ease Messen-ger	March Networks™ 6500 Speech Enabled Unified Messaging	March Networks 6500 Speech Enabled Unified Messaging	6500 SE Unified Messag-ing
7XXX Applications Network Management			
OPS Manager for 3300 ICP	March Networks™ 7100 Network Management	March Networks 7100 Network Management	7100 NM
8XXX Business Services			
8100	March Networks™ 8100 Service Solutions Portfolio	March Networks Service Solutions Portfolio	Service Solutions Portfo-lio
8200	March Networks™ 8200 Managed Broadband Services	March Networks Managed Broadband Services	Managed Broadband Ser-vices

NOTES