MITEL NETWORKS

3300 | Integrated Communications Platform

GENERAL INFORMATION GUIDE

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Overview

The Mitel Networks™ 3300 Integrated Communications Platform (ICP) is a Voice over IP (VoIP) solution that delivers the superior voice capabilities and the features that enterprises have come to expect from a communications system. The 3300 ICP builds upon Mitel Networks 25+ years of voice experience, migrating system and desktop features forward into a packet-switched delivery model. Tailored for small, medium, and large enterprises, the Mitel Networks 3300 ICP provides a complete solution for all communication needs by delivering sophisticated call management, desktop peripherals, and applications.

Mitel Networks solution is comprised of control elements, media gateway elements, a rich portfolio of peripherals and applications including Mitel Networks IP telephones, wireless IP telephones, and Mitel Networks digital telephones. Applications include voice mail, speech-enabled auto attendant and unified messaging, as well as a full line of call center options all easily managed by web based applications. The system also supports a wide range of analog and digital trunks for connection to the Public Switched Telephone Network (PSTN), and for connecting multiple sites or systems together. In addition the 3300 ICP provides IP Networking enabling multiple sites to be connected via the LAN/WAN infrastructure.

The 3300 ICP provides fully featured call control services, functionality and applications associated with a traditional PBX such as multiple levels of call forwarding, message waiting, advisory messages, conference calling, account codes, call barring, least cost routing, night service plus 100's more. The 3300 ICP delivers the value of VoIP through integrated applications and converged infrastructure that provide a lower total cost of ownership.

The 3300 ICP can operate on a stand-alone basis and has the ability to cluster multiple systems over IP- or TDM-based infrastructures - thus supporting thousands of onsite or remote users. All of these clustered systems can all be managed as a single network from any authorized location.

Network Solutions
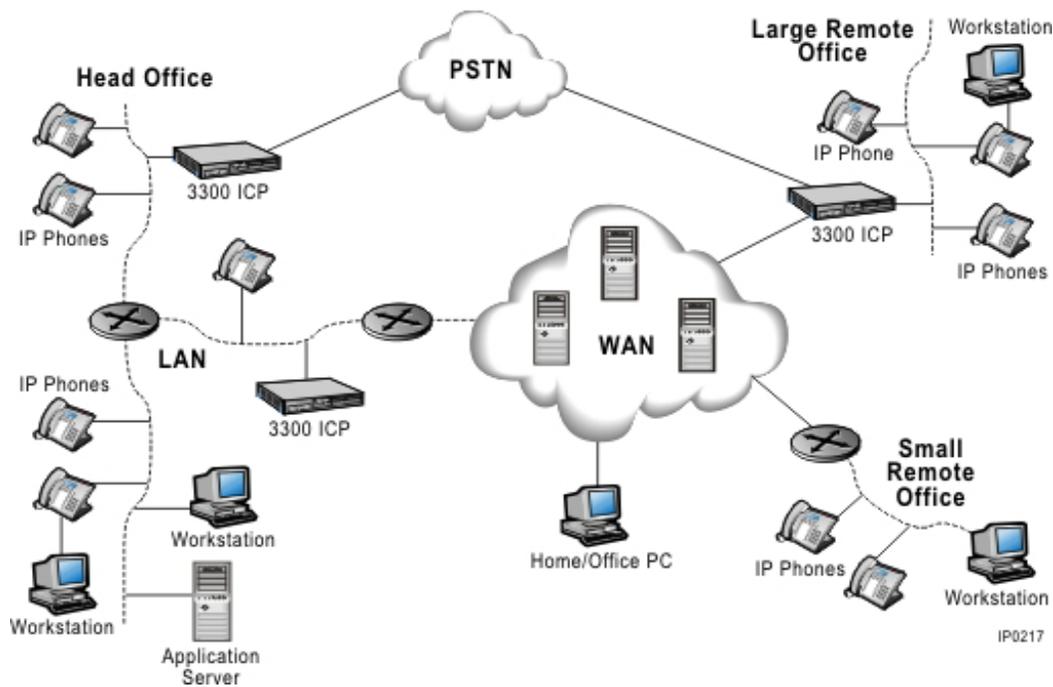
Refer to the Solutions section of this document for examples of different system configurations.

System Architecture

The 3300 ICP is built upon Mitel Networks Data Integrated Voice Applications™ architecture delivering sophisticated call management, applications and desktop solutions for businesses. Mitel Networks delivers a highly scalable robust call control that fully utilizes the power of IP while recognizing the need to fully support traditional TDM based telephony for legacy devices and PSTN connectivity.

Mitel Networks' architecture uses the IP network for connecting IP telephony devices and provides a supplementary TDM (Time Division Multiplexing) subsystem for switching calls between traditional telephone devices. The 3300 ICP has the advantage of being able to optimally switch all types of traffic, IP or TDM. The 3300 ICP provides native call setup, tear down, and signaling between Ethernet IP connected telephones. For traditional telephony, such as POTS and PSTN trunks, call handling is also handled natively by the 3300ICP via a conventional TDM circuit-switched subsystem.

This ability to use two different switching techniques simultaneously means that

- All traffic is switched with minimum conversion between packet and traditional telephony to provide optimum voice quality in all call scenarios.
- Embedded gateway functionality is only required between the IP and non-IP networks optimizing the use of system resources.
- Migration from traditional PBX to IP telephony is seamless and efficient.
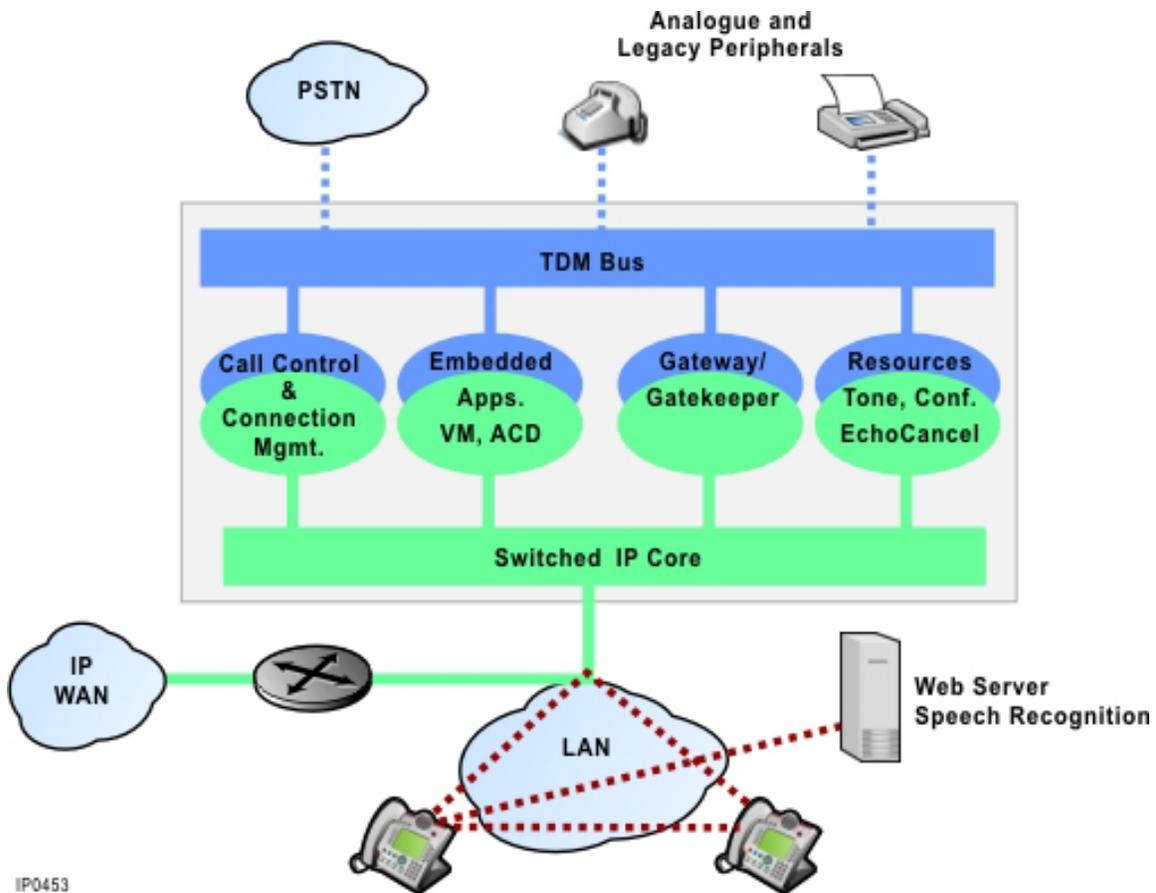
System

Physical System Features

The 3300 ICP components have the following common physical features:

- External Casing. All of the components may be stacked or rack-mounted (in a 19-inch rack)
- Power Supply. Each unit has its own Standard Male IEC AC input connector for power
- LEDs. All LEDs are located on the front of the units for visual indication of circuit status.

Controller

The Mitel Networks 3300 Controller provides the voice, signaling, central processing, and communications resources for the system.

The 3300 Controller contains a hard drive, three RS-232 ports (DB-9 connectors), a remote alarm port, 64 or 128 channels of echo cancellation, up to two dual FIM modules, up to four 10/100 BaseT Ethernet Ports, up to four CIM Expansion Ports, a Stratum 3 clock, and a power-fail-protected real-time clock.

The 3300 Network Services Units (NSUs) connect to the 3300 Controller using multi-mode fiber terminated on an ST connector using the dual FIM module located on the front panel.

The 3300 Analog Services Units (ASUs) connect to 3300 Controller using copper cables terminated on an RJ-45 connector using the CIM expansion ports located on the front panel.

IP devices connect to the customer's LAN which in turn connects to the controller using the four 10/100BaseT Ethernet switch ports located on the front panel.

In addition, SX-2000 Digital Services Units (DSUs) and Peripheral units can be connected over a fiber cable through a FIM module.

Three sizes of 3300 Controller are available. These units provide direct connectivity for the following maximum configurations

- 100 users, 2 NSUs direct connection (4 NSUs if chaining is used), 2 ASUs, 8 Attendant Consoles, 64 echo cancellers, and 32 G.729a compression channels (optional). Note: This configuration can not be expanded.

- 250 users, 2 NSUs direct connection (4 NSUs if chaining is used) 4 ASUs, 16 Attendant Consoles, 64 echo cancellers, and 32 or 64 G.729a compression channels (optional). Note: This configuration may be expanded up to 700 users.

- 700 users, 4 NSUs direct connection (8 NSUs if chaining is used), 4 ASUs, 16 Attendant Consoles, 128 echo cancellers and 32 or 64 G.729a compression channels (optional).

📝 Note: A peripheral unit or DSU connected to the controller will use a FIM and thus reduce the number of NSUs that can be connected.

For specific details on the 3300 Controller hardware, refer to the "Hardware User Guide."

3300 Controller (700 users)

Network Services Units

The Network Services Unit provides connectivity to digital trunks for public or private networks. Protocol support includes DASS II, PRI, BRI, MSDN/DPNSS, R2, T1/ CAS, Q.Sig, and XNET.

There are three variants of NSU:

* Mitel Networks 3300 Universal Network Services Unit
* Mitel Networks 3300 R2 Network Services Unit
* Mitel Networks 3300 BRI Network Services Unit

Mitel Networks 3300 Universal Network Services Unit

The Mitel Networks 3300 Universal NSU provides T1 or E1 connectivity and supports up to two T1 or E1 links per unit. The protocols supported by the T1 interfaces are

* T1 CAS - Digital E&M, Digital CO, Digital DID
* T1 CCS - Primary Rate ISDN (4ESS, 5ESS, DMS 100, DMS 250, NI2, NI3), XNET over PRI, Q.Sig and MSDN/DPNSS.

The protocols supported by the E1 interface are:

* Q.Sig, Euro ISDN, XNET over PRI, DASS II, and MSDN/DPNSS

The 3300 Universal NSU connects to a 3300 Controller through a fiber cable.

Additional digital trunk capacity can be added to the 3300 ICP by chaining two NSUs together via the Copper Interface Module (CIM) connection using a Category 5 crossover cable.

For detailed information on the hardware specifications, refer to the "Hardware User Guide."

Mitel Networks 3300 R2 Network Services Unit

The Mitel Networks 3300 R2 NSU provides connectivity to R2 trunks with MF-R2 digital trunk signaling. Up to two links are supported by a single 3300 R2 NSU.

The 3300 R2 NSU connects to a 3300 Controller through a fiber cable.

Additional digital trunks can be added to the 3300 ICP by chaining two NSUs together via the Copper Interface Module (CIM) connection using a Category 5 crossover cable.

For detailed information on the hardware specifications, refer to the "Hardware User Guide."

Mitel Networks 3300 BRI Network Services Unit

The Mitel Networks 3300 BRI NSU provides connectivity for Basic Rate ISDN (BRI) transport for both data and voice traffic. This unit is available in North American and European variants. The North American variant supports user-side interface. The European variant supports both network and user-side interfaces.

The 3300 BRI NSU supports 15 BRI U-interfaces per unit. It does not connect directly to a 3300 Controller but rather to the Mitel Networks 3300 Universal NSU through a copper cable. For detailed information on the hardware specifications, refer to the "Hardware User Guide."

Analog Services Units

The Analog Services Unit (ASU) provide connectivity for analog trunks and telephones (POTS and On-Premise Station (ONS)) to the 3300 ICP system. There are two variants:

- Mitel Networks 3300 Universal Analog Services Unit
- Mitel Networks 3300 Analog Services Unit

The 3300 ICP system supports up to four Analog Services Units in any combination giving a maximum of 96 analog ports for standard telephone connectivity.

Mitel Networks 3300 Universal Analog Services Unit

The Mitel Networks 3300 Universal ASU is a combination unit that houses 16 ONS Custom Local Area Signaling System (CLASS) and four Loop Start (LS) trunk ports. It also provides four integral System Fail Transfer (SFT) relays that provide direct connection between an analog telephone and Loop Start trunk in the event of a system or power failure.

The 3300 Universal ASU also provides the connections for Music on Hold (MOH) and Paging capabilities. The connections are located on the rear panel. The unit connects to a 3300 Controller by using a Category 5 Universal Twisted Pair (UTP) cross-over cable that terminates on an 8-pin modular jack (RJ-45).

Note: CLASS is not currently supported on the UK or European variants of this unit.

For detailed information on the hardware specifications, refer to the "Hardware User Guide."

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Mitel Networks 3300 Analog Services Unit

The Mitel Networks 3300 ASU is the On-Premises Station (ONS) Line unit and supports up to 24 ONS CLASS ports. It connects to a 3300 Controller by using a Category 5 UTP cross-over cable that terminates on an 8-pin modular jack (RJ-45).

Note: CLASS is not currently supported on the UK and European variants of this unit.

For detailed information on the hardware specifications, refer to the "Hardware User Guide."

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Digital Service Unit

A DSU cabinet holds up to four DSU cards and two FIMs. The DSU cards provide digital trunk interfaces for public or private network access and specialized digital functions (such as messaging and ISDN service). The FIMs support the transmission of voice and data signals between the DSU and the 3300 controller.

DSU cards include:

- DS1 Formatter
- CEPT Formatter
- E1/T1 Digital Trunk Formatter
- BRI card (6 circuit)
- BRI card (15 circuit)
- Euro PRI card
- NA PRI card
- R2 card.

For detailed information on the hardware specifications, refer to the 3300 ICP "Hardware User Guide."

Peripheral Unit

Each peripheral unit holds up to 12 peripheral interface cards and provides up to 192 ONS or DNI ports. A slave cabinet can be added that expands the unit 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) are installed in the master cabinet of each peripheral unit. The PSC card provides control for all peripheral interface cards, and a fiber-optic cable connects the FIM to the 3300 controller.

The following peripheral interface cards are available:

- DNI Line card
- LS/GS trunk card
- ONS CLASS/CLIP Line card
- ONS line card
- E&M Tie trunk card
- OPS line card
- DID/Loop Tie trunk card.

Note: Not all cards listed are supported in all countries.

For detailed information on the hardware specifications, refer to the 3300 ICP "Hardware User Guide."

Migration

The 3300 ICP offers the ability to migrate from an existing SX-2000 MICRO LIGHT, SX-2000 LIGHT, or 3200 ICP. The Mitel Networks 3300 Configuration Tool converts and restores the existing

database to the 3300database is converted and restored onto the 3300 ICP by using the Mitel Networks 3300 Configuration Tool.

Mitel Networks SX-2000 MICRO LIGHT

By installing a triple FIM card in the SX-2000 MICRO LIGHT cabinet, you can connect it to the 3300 ICP with a multi-mode fiber cable. As a result, you can use the existing peripheral and digital trunk cards within the cabinet. Any additional cabinets can also be connected in this way.

Mitel Networks SX-2000 LIGHT

The SX-2000 LIGHT Digital Services Unit (DSU) cabinet provides digital trunk capability, and the peripheral cabinet provides connectivity for analog trunks, analog telephones, and Mitel Networks DNI devices. Both cabinet types can be connected to the 3300 Controller through multi-mode fiber connections.

The DSU cabinet supports BRI, PRI, T1/D4, MSDN/DPNSS, and DASS II trunks.

The peripheral cabinet supports the following analog trunks:

- Analog CO trunks
- E&M trunks
- Direct-inward-dial and tie trunks.

It also supports the following DNI telephones and devices:

- Superset™ 401
- SUPERSET 401+
- SUPERSET 410
- SUPERSET 420
- SUPERSET 430
- SUPERSET 4001
- SUPERSET 4015
- SUPERSET 4025
- SUPERSET 4125
- SUPERSET 4150
- Superconsole™ 1000
- Mitel Networks Analog Interface Module.

For additional information, refer to SX-2000 technical documentation.

Mitel Networks 3200 Integrated Communications Platform (ICP)

The Mitel Networks 3200 ICP database is converted and restored to a 3300 ICP database, and any peripheral cabinets connect to the FIM ports on the 3300 Controller.

In addition to supporting DNI telephones with the peripheral cabinet, the 3300 ICP supports the following legacy, single-port IP telephones:

- SUPERSET 4015IP
- SUPERSET 4025IP.

Mitel Networks 3800 Wireless Applications Gateway

You can migrate the Mitel Networks 3800 Wireless Applications Gateway database to the 3300 ICP, using the in-built OPS manager Application and the 3300 Configuration Tool.

Maintenance

Alarms

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

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

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

Circuit Indicators

The front of each system component has light emitting diodes (LEDs) that indicate the status of the power, trunk circuits, line circuits, message links, and alarm status (as applicable).

For details, refer to the "Hardware User Guide."

System Management Tools

The system has the following programming tools that have been designed for different levels of user:

- The System Administration Tool provides a Web-based interface that trained technicians use to program the system. It requires Microsoft Internet Explorer 5.5 or higher.
- The Group Administration Tool provides a Web-based interface that enables administrators and receptionists to make changes to user information. It requires Microsoft Internet Explorer 5.5 or higher.
- The Desktop Tool provides a Web-based interface that enables IP telephone users to program their telephone feature keys on their phone. It requires Microsoft Internet Explorer 5.5 or higher.

In addition the following tools are available to assist with the configuration of the system.

- The Mitel Networks 3300 Configuration Tool enables the installer to get a new system up and running at the installation site. It also enables databases from legacy SX-2000 Light, SX-2000 MicroLight systems, 3200 ICP systems, and 3800 WAG systems to migrate to the 3300 ICP through a database conversion and restoration utility. The 3300 Configuration Tool requires Microsoft Windows® NT 4.0 or Microsoft Windows® 2000 Professional operating system.
- The ISDN Maintenance and Administration Tool (IMAT) provides the programming interface for PRI and R2 protocols. It requires Microsoft Windows® 95, Microsoft Windows® 98, or Microsoft Windows® 2000 Professional.

Redundancy Support

The 3300 ICP can auto-fail-over (route) around failed IP links if provisioned redundantly.

Security

Toll Control

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

The 3300 ICP system has comprehensive toll control as an integral part of the call control. It lets you restrict user access to trunk routes and/or specific external directory numbers. It also provides Class of Restriction (COR) and Class of Service (COS) features that can substantially reduce the risk of toll abuse.

Features that have a risk of toll abuse are

- Public Network to Public Network Connection Allowed - permits or restricts trunks the connection of trunks.
- Call Forwarding External Destination - allows or restricts extension user to forward calls to external trunks
- Automatic Route Selection - allows or restricts, on a per user or system basis, access to directory numbers based on a users job function (note: 1-800 calls are usually free calls, but some central offices can allow the reversal of 800 charges so that they are toll calls for your company).

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

Authorized Access

Authorized access to the system tools provides protection for various administration commands from unauthorized users. The Web-based system tools are

- System Administration Tool
- Group Administration Tool
- Desktop Tool.

Each user is given a login name, password, extension number, and language preference. All systems should have all levels of passwords and login names altered from the default value; moreover, these passwords should be changed periodically.

Ensure that any voice mail systems connected directly to modems employ a surveillance device. Also, since most voice mail systems require a password to gain access, make sure that this password is difficult to guess and is changed frequently. Any user no longer authorized to use the system should have his/her password privileges revoked immediately.

Specifications

Environment for 3300 ICP Components and Peripherals

Storage Environment for 3300 ICP Components and Peripherals	
Condition	**Specification**
Temperature	3300 Controller: 32º to 104º F (0º to 40º C)
	All Other Components: 39º to 120ºF (4º to 49ºC)
Humidity	15-95% Relative Humidity, non-condensing
Vibration	0.5 g, 5 to 100 Hz, any orthogonal axis
	1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress	One 15.3 cm (6 in.) drop, each edge and corner adjacent to the rest face – unpackaged
	One 76.2 cm (30 in.) drop, each edge and corner packaged in cardboard & foam.

Operational Environment for 3300 ICP Components and Peripherals	
Condition	**Specification**
Temperature	3300 Controller: 59º to 95º F (15º to 35º C)
	All Other Components: 39º to 120ºF (4º to 49ºC)
Humidity	3300 Controller: 40-90% Relative Humidity, non condensing
	All Other Components: 34-95% 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
Acoustic Emissions	Maximum 50 dBA continuous, 75 dB intermittent (<10% duty cycle)
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.	

Environment for DSU and Peripheral Units

Storage Environment for DSU and Peripheral units	
Condition	**Specification**
Temperature	-40º to 150ºF (-40º to 66ºC)
Humidity	5-95% Relative Humidity, non-condensing
Vibration	0.5 g, 5 to 100 Hz, any orthogonal axis
	1.5 g, 100 to 500 Hz, any orthogonal axis
Mechanical Stress	One 20.3 cm (8 inch) drop, each edge and corner adjacent to the rest face

Operational Environment for DSU and Peripheral units	
Condition	**Specification**
Temperature	32º to 122ºF (0º to 50ºC)
Humidity	5-95% Relative Humidity, non-condensing
Maximum Heat Dissipation - fully loaded (see Note)	Peripheral Cabinet 724 BTUs per hour DSU Cabinet 266 BTUs per hour
Air Flow	150 cubic feet per minute at maximum output of the fans
Acoustic Emissions	Maximum 50 dBA continuous, 75 dB intermittent (<10% duty cycle)
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.	

Additional information can be found in the 3300 ICP "Hardware User Guide."

Dimensions and Weights

Component	Height	Width	Depth	Weight
3300 Controller	2.75 in. (7 cm)	19.0 in. (48.3 cm)	15.5 in. (39.4 cm)	16.19lbs (7.35kg)
3300 Universal NSU	1.75 in. (4.45 cm)	19 in. (48.3 cm)	15.5 in. (39.4 cm)	9.25lb (4.2kg)
3300 R2 NSU	1.75 in. (4.45 cm)	19 in. (48.3 cm)	15.5 in. (39.4 cm)	9.63lb (4.37kg)
3300 BRI NSU	1.75 in. (4.45 cm)	19 in. (48.3 cm)	15.5 in. (39.4 cm)	9.57lb (4.34kg)
3300 Universal ASU	1.75 in. (4.45 cm)	19 in. (48.3 cm)	15.5 in. (39.4 cm)	10.54lb (4.79kg)
3300 ASU	1.75 in. (4.45 cm)	19 in (48.3 cm)	15.5 in (39.4 cm)	9.98lb (4.53 kg)
Digital Service Unit	19 in. (48.3 cm)	9 in. (22.9 cm)	16.5 in. (42 cm)	40lb (18 kg)
Peripheral Cabinet	19 in. (48.3 cm)	18 in. (48.3 cm)	16.5 in. (42 cm)	95lb (43.2 kg)

Tone Plans

The system supports tone plans for the following countries:

- Australia
- Brazil
- France
- Germany
- Italy
- Latin America (Argentina, Chile, Mexico)
- Netherlands
- New Zealand
- North America (Canada, USA)
- Portugal
- Spain
- United Kingdom.

For detailed information, refer to the technical information in the "Hardware User Guide."

Applications

System Applications

Applications Interfaces

The following application interfaces are supported on the 3300 ICP system:

- TAPI™ - Microsoft's TAPI (Telephony Application Programming Interface) for desktop applications or client/server applications.
- MiTAI - The Mitel Telephony Applications Interface (MiTAI) is an Applications Protocol Interface (API) that allows third-party-developed CTI applications to interface with the Mitel Network's call control. A programmer's toolkit plus run-time software is also available, which enables developers to create computer telephony applications.

For additional information refer to "Mitel Networks Enabling Technology" documentation.

Automatic Call Distribution

The 3300 ICP provides fully integrated Automatic Call Distribution (ACD) functionality that includes call distribution, agent mobility, management and reporting, feature configuration, administration, and recorded announcement devices (RADs).

The system also supports ACD over the MSDN/DPNSS digital network or IP to allow multi-site working; agents at different locations service calls on the network independently of where they entered the network.

Additional functionality can be added with the Mitel Networks 6100 Contact Center Solutions product line.

Integrated Wireless

The 3300 ICP supports wireless devices from Symbol Technologies to provide a fully-featured integrated wireless solution. With Symbol Technologies wireless IP phone devices, users have access to the complete range of 3300 ICP features over an 802.11b wireless LAN. The 3300 ICP supports the Symbol MiNET Wireless Phone (NetVision II).

Using the IEEE 802.11b wireless LAN protocol a network of Symbol Spectrum24 Access Points form the basis of the wireless LAN. These Access Points function as a Media Access Control (MAC) bridge between wired LANs and wireless endpoints. The Spectrum24 Access Points use the Direct Sequence Spread Spectrum and operate at up to 11 Mbps.

The 3300 ICP can be used to provide wireless gateway functionality to existing legacy PBX's including Mitel Networks SX-2000 and Mitel Networks SX-200 systems. Communication between the 3300 ICP wireless gateway and the legacy PBX is over standard trunk protocols such as MSDN/DPNSS, Q.Sig, or T1/D4.

For more information on Symbol Technologies products visit their Web-site at www.symbol.com.

Voice Mail

The 3300 ICP includes an integrated fully featured voice mail system designed to improve communication among your company, clients, customers, and employees. Up to 30 ports are available for voice mail calls with support for a maximum of 750 mailboxes and 450 hours of storage time. The voice mail system supports English, European French, Canadian French, European Spanish, Latin American Spanish, Dutch, German, Italian, Brazilian Portuguese and European Portuguese.

Features provided by the voice mail system include:

- An automated attendant that plays different greetings during open and closed business hours, provides a company directory that uses extension numbers or names as the dialing method, and allows single-digit option selection
- Multi-level auto attendant allows a hierarchical menu to be programmed on the auto attendant providing callers with better self-service access to the person or department they are calling.
- User mailboxes that are password-protected
- A tutorial that assists new subscribers with mailbox setup
- Simple message retrieval
- Easy-to-use menus that allow subscribers to send urgent, private, or certified messages
- Notification of waiting messages whether subscribers are in or out of the office.

For a complete listing of features provided by the 3300 ICP voice mail refer to the Auto Attendant Features list and the Voice Mail Features list.

Capacities

Voice mail ports:	Up to 20 (programmable) with base offering 30 (optional) see details in the "Hardware User Guide."
Mailboxes:	750
Disk space for voice mail files	14GB
Hours of voice storage:	450
Concurrent voice mail or auto attendant sessions	30
Message storage per mailbox	100 maximum (programmable)
Message retention	From one day to indefinitely for saved messages; indefinitely for unread messages. (Programmable on a per mailbox basis).
Prompt languages:	English, European French, Canadian French, European Spanish, Latin American Spanish, Dutch, German, Italian, Brazilian Portuguese and European Portuguese

External voice mail systems are supported through both digital trunk and ONS interfaces.

Management Applications

OPS Manager

OPS Manager is a complete telecommunications management tool that enables customers to control the maintenance and operation of a network of Mitel Networks systems. Using a standard web browser, an authorized user can perform the following functions from any PC:

- Manage a network telephone directory
- Move, add, change, and delete users
- Schedule pending moves, adds, and changes to the 3300 ICP database
- Integrate the network telephone directory with a network directory service database
- Monitor alarm-status messages that are automatically reported from the network
- Audit the status of the managed devices
- Perform remote programming and maintenance
- Locate unused directory numbers and unused circuits.

OPS Manager is available as software only (installed on a user/dealer provided server).

Note: OPS Manager is a Java™-based application that supports multiple client stations; therefore, you can access the application through a Netscape® Communicator 6.2 browser or a Microsoft Internet Explorer 5.5 sp4 browser from any Windows NT, Windows 95/98, or Windows 2000 workstation on the network.

Mitel Networks 6200 Cost Management Solution

Mitel Networks 6200 Cost Management Solution (CMS) is the application portfolio of choice for monitoring and managing communication resources. Mitel Networks 6200 Cost Management Solution is a fully-integrated voice and data network management solution. Helping customers to monitor their voice & data resources and empowers them to make good decisions about their communications network.

Count on the 6200 Cost Management Solution to deliver:

- Maximum ROI for voice and/or data communications
- Reduced operations costs through identification of:
- Redundant services
- Need to up (or down) grade system hardware
- Misuse/neglect of voice and data resources

The 6200 Cost Management Solution is broken into two main categories:

- Professional – Targeted at customers with less than 3500 stations. All modular options within the Professional categories are bundled in packages of 50.
- Enterprise – Targeted at customers with more than 3500 stations. All modular options within the Professional categories are bundled in packages of 1000.

Each category within the 6200 Cost Management Solution consists of the following modular options:

Call Accounting

- Supports all carrier billing structures
- Accurate costing and call usage details
- Many standard report/sort types/filters available
- Flexible, customizable retrieves data from OPS Man or the 6100 CCM

General Cost Allocation

- Third party billing and costing application
- Flexible import capability to introduce third party billing data directly into the 6200 CMS
- Supports most data formats including ASCII, Excel, Access, and any ODBC data source
- Assign costs to appropriate chargeable entities for a total overview of costs

Internet Usage Reporting

- Monitor employee Internet behavior including:
- Where (sites)
- What (content)
- When (time of day and duration)

Email Usage Reporting

- Provides effective management and reporting capabilities for all email usage within your company
- Manage employee email usage by tracking:
- Source and Destination
- Date
- Size

Traffic Reporting

- Offers reports on trunk utilization to help customers determine if their organization is under- or over-trunked.

For detailed information refer to Mitel Networks 6200 Cost Management Solutions documentation.

Mitel Networks 6300 Call Recording Solutions

The Mitel Networks 6300 Call Recording Solutions provide customers with the tools that they need to record calls for liability, compliance, auditing, or order/transaction verification. It also allows evaluation of the quality of agents' work by measuring performance so supervisors can help agents provide the best possible service.

Each recording product in the 6300 Call Recording Solution (PRO, ADVANCED and, TDI) provides specific capabilities, ease of use and unprecedented flexibility. Mitel Networks provides solutions for recording four to thousands of extensions. All of these systems provide extension side recording for analog, digital, and IP sets or trunkside recording on PRI NI-2 or PRI 4ESS.

	Core Features	Optional
MARATHON Advanced	- 4-64 channels analog - 4-32 channels Mitel Digital - TCP/IP - Standard 1,000-hour hard disk - Call Collection Box for storage of archive calls to hard disk for instant retrieval - Keyboard, mouse, 17" Monitor Integrated Speakers - Loop Playback facility - Remote diagnostics and remote service - Record Mode: Cascade, Sequential or Parallel. - Compression Rate: 15,24,32,40,64 Kbit/sec, selectable per channel Simultaneous record and playback from the active archive media.	- CTI integration for free seating and precise search by agent ID, Transaction Code - Selection of conversations according to events, scope of business, or application - Extended storage capacity: 2,000 or 4,000 hours - Redundant Hard Disk - Archive Media: 0, 1, or 2 DAT (DDS-1, 2,3,4), MOD or DVD Drives
MARATHON Pro	- 8-128 channels analog - 4-64 channels Mitel Digital - Can support thousands of networked channels - TCP/IP - Standard 1,000-hour hard disk - Loop Playback facility - Remote diagnostics and remote service - Simultaneous record and playback from the active archive media - Record Mode: Cascade, Sequential or Parallel. - Compression Rate: 15,24,32,40,64 Kbit/sec, selectable per channel	- Media Storage Tray with up to 10,000 channel hours of unattended recording - Can be connected to media library for unlimited storage - Fault Tolerant design including: redundant components or complete hot stand-by system - CTI capability with individual security level, and ability to search by: user name, log on ID, call direction - Selection of conversations according to events, scope of business, or application - Redundant Hard Disk - Extended storage capacity: 2,000 or 4,000 hours - Redundant load sharing power supply - Extended storage capacity Archive Media: 0, 1, 2 or 3 DAT (DDS-1, 2,3,4), MOD or DVD Drives

	Core Features	Optional
MARATHON TDI	- Recording, buffering, and automatic data transfer to a central file server system - Channel capacity up to 128 per chassis (64 for Mitel Digital) - Data Storage and buffering on the local hard disc (up to 4,000 hours), call related data stored in local database - Optional RAID mirroring of local hard disc - Automatic up-load—redundancy of files for security - CTI and VOX controlled recording, agent or supervisor initiated record on demand, event-driven, or rules-based recording(e.g. When this incoming number appears record, if more than 'x' transfers or more than 'x' hold time capture it). - Search and Replay by browser-based applications (Microsoft Internet Explorer) - Embedded operating system - Open standard SQL-database concept - Microsoft SQL server - Compatible with all telephone systems and dealer boards, - Connectivity - Trunk side: PRI NI-2 or PRI 4ESS - Digital DNIC recording - Analog - Tap card for proprietary digital subscriber lines;	-

Optional Management Features for the Marathon PRO & Advanced

- Client Workstation Software Provide central remote control for the Marathon Advanced & Marathon Pro (Control for up to 24 Marathon recorders) - Flexible Search & Replay - Live monitoring - Recorder set-up & configuration - Easy to use GUI

- Instant Recall Terminal The Instant Recall Terminal (IRT) is a remote control application, allowing quick and easy access to conversations recorded on the MARATHON. The purpose of this application is to immediately play back the last call in an emergency situation. Offered in a hardware and software version, the unit is soft key driven and equipped with a large display to provide necessary information to the operator, along with the recording.

- Last Call Repeat The optional Last Call Repeat Facility enables up to 8 users quick access to their last calls from their own telephone. This enables a user to monitor recorded conversations from any telephone throughout the world, 24 hours a day, without relying on any administrative personnel.

- Threat Call Recording This software package in combination with the MARATHON recording system allows selective recording of important calls and threat calls. The software is installed on each corresponding PC workstation that can be associated with a specific telephone. The calls are selected by a mouse click and automatically transferred to a call collection box. From there critical calls can be accessed and played back.

Optional Management for the Marathon TDI

- Search & Query via Low-Bandwidth (WWW) connections

- Features (Webplayer 2.1)

- Truly Browser Based – No Applet's need to beloaded

- Quick Access – fast query & replay

- Access Control via Login,

- Quick Search (by time, channel, call comment, CLI, Agent ID)

- Detailed Search (by all DB-fields)

- Last Call(s) Repeat – quick playback of the most recent calls

- Save recording as Wav

- Add and save Call Comment

- Multi Language- Time-Format Support (English UK, USA, German)

- Quality Assurance (browser-based application overlays recording platform) – See the 6170 Quality Monitoring Solution for additional product details.

Mitel Networks 5700 Voice First Application

Mitel Networks 5700 Voice First Application is a software-based IP video system. It provides the easiest-to-use video conferencing solution on the market today by integrating it's application with the controls on a Mitel Networks 5020 IP Phone. When combined with the Mitel Networks 5750 Desktop Video Appliance or 5760 Video Conference Appliance, the 5700 Voice First Application provides a powerful video conferencing solution that enables companies to eliminate travel costs and set up effective communications between remote offices.

This application should be installed on an NT server with the following minimum requirements:

- 400 MHz Pentium
- 128 MB Memory
- 10/100 Network Interface card
- Windows NT or Windows 2000 operating system.

Mitel Networks 5700 Voice First Application Billing & Reporting

Mitel Networks 5700 Voice First Application Billing and Reporting provides a billing functionality that correctly allocates costs. This optional module provides SMDR like records for up to 100 video end points in a network. Adding more modules enables the monitoring of more end points. The 5700 Voice First Application Billing & Reporting Module resides on the same NT server as the 5700 Voice First Application.

The following pre-formatted reports are included with the 5700 Voice First Application Billing and Reporting:

- Dial plan report
- Gateway report
- MCU report
- User reports
- System activity Report
- Call frequency report
- Call details report.

Note: This application should be installed on the same server as the 5700 Voice First Application.

Mitel Networks 5720 Voice First Application Conference Bridge

Mitel Networks 5720 Voice First Application Conference Bridge enhances the 5700 Voice First Application by allowing additional participants to take part in a video conference with the touch of a button. A single session can be added to the same server as the 5700 Voice First Application. Once additional sessions are required beyond a single session it is best to use a separate server for the voice conference bridge (VCB). Additional features include:

- 1,2,4 or 8 session configurations (separate PC's are required for 2,4, & 8 configurations)
- Continuous Presence
  - Up to 384 Kbps
  - Up to 5 participants per session
- Voice Activated Switching
  - Up to 1.5 Mbps
  - Up to 8 participants per session

- Dial in or Ad-hoc conferencing.

This application should be installed on a PC with the following minimum requirements:

- 400 MHz Pentium
- 64 MB Memory
- 10/100 Network Interface card
- Windows NT or Windows 2000 operating system.

Mitel Networks 5750 Desktop Video Appliance

The Mitel Networks 5750 Desktop Video Appliance is a personal appliance for use with both desktop and laptop configurations, enabling users to make direct calls to other personal conferencing systems and to participate in multipoint meetings.

Features of this appliance include:

- IP conferencing rates of up to 1.5 Mbps
- Call Transfer and Call Forward features (available with 5700 Voice First Applications)
- Data Sharing, File Transfer, and White-board
- Interactive Multicast technology
- Centralized Management and Administration (with 5700 Voice First Applications).
- Enhanced Quality of Service.

The Mitel Networks 5750 Desktop Video Appliance is available in three options: Standard, Professional, and Executive.

Note: A Mitel Networks 5020 IP phone is required to provide the audio communication for the 5750 Desktop Video Appliance.

Mitel Networks 5760 Video Conference Appliance

The Mitel Networks 5760 Video Conference Appliance is a desk-top appliance that operates over both IP and ISDN networks.

Features include:

- Compact industrial design
- IP data rates up to 512Kbps
- T.120 data Sharing over ISDN
- Enhanced Quality of Service
- Interactive Multicast Participant
- Web-based management
- Call Transfer and Ad-hoc conferencing with the use of 5700 Voice First Application.

The Mitel Networks 5760 Video Conference Appliance is available in Standard and Enhanced versions.

Note: A Mitel Networks 5310 IP Board Room Conference Unit is required to provide the audio communication for the 5760 Video Conference Appliance.

Mitel Networks 6100 Contact Center Solutions

Mitel Networks 6100 Contact Center Solutions (CCS) is a suite of applications that enhances the integrated ACD functionality of the 3300 ICP. These applications enable customers to maximize the efficiency of their contact center. This suite comprises

- Mitel Networks 6110 Contact Center Management (CCM)
- Mitel Networks 6115 Interactive Contact Center
- Mitel Networks 6120 Contact Center Scheduling
- Mitel Networks 6150 Multimedia Contact Center (MCC)
- Mitel Networks 6160 Intelligent Queue (IQ)
- Mitel Networks 6170 Quality Monitoring
- Mitel Networks 6200 Cost Management in a Contact Center

Mitel Networks 6110 Contact Center Management

Mitel Networks 6110 Contact Center Management (CCM) is a browser-based contact-center-management solution that provides supervisors and managers with advanced capabilities for

- Forecasting contact center activities based on historical activities and what-if scenarios
- Real-time agent/supervisor communications
- Real-time monitoring of contact center activities, including queue and agent status, average wait times, and more
- Viewing of historical events
- Historical reporting including On the fly, and scheduled report generation on all contact center elements and contact types over any date and time span

The 6110 CCM includes the following features:

- Real-time Monitoring –SuperAdvisor
- Historical Monitoring –SuperAuditor
- Historical Reporting -Reporter
- Forecasting –4Sight
- Agent real-time Monitoring –AgentAdvisor
- Instant Messaging –ChatLine
- Spectrum Wallboard support –Wallboarder
- ACD & SMDR Search capabilities –Inspector
- Multi-site historical reporting, real-time monitoring & agent forecasting –Enterprise Node

Mitel Networks 6115 Interactive Contact Center

The Mitel Networks 6115 Interactive Contact Center provides supervisors with browser-based tools that enable them to:

- Dynamically control the availability of individual agents and queues
- Schedule ACD queues to automatically open and close according to business hours
- Enable Do Not Disturb on specific queues, diverting callers to alternative answering points
- Make individual agents busy with reason codes
- Communicate with agents when their state has changed.

Mitel Networks 6120 Contact Center Scheduling

Mitel Networks 6120 Scheduling integrates with the 6110 CCM forecasting to provide supervisors with automated capabilities to

- Automatically schedule agents to meet forecasted contact center activity levels on a per shift, daily, weekly and monthly basis
- Efficiently schedule agents by assigning them tasks based on their skill set
- Schedule within budget by viewing the automatically generated reports on the financial impact of each schedule
- Adjusting schedules on the fly when conditions change
- Plan and track training, meetings, vacations, and sick leave with sophisticated accrual-based leave planning
- Storing employee skills, hire dates, payroll rates, ID photos, addresses, and emergency information in employee profiles
- Reporting on schedules, budgets, or payroll rates–there are 29 reports in all
- Importing or exporting scheduling information using dBase, Microsoft Access, Excel or FoxPro.

Mitel Networks 6150 Multimedia Contact Center

Mitel Networks 6150 MCC is a modular offering that integrates with Microsoft Exchange 2000 and Outlook to

- Automatically route incoming email and Web-chats to the next available agent
- Create unique auto-acknowledgement messages based on email aliases and email queues
- Assign a unique tick number to each incoming and outgoing email, linking the customer's entire email history together.
- Provide supervisors with the ability to view, generate, schedule, and share reports across all contact center multimedia elements over any date & time horizon.
- Provide supervisors with dynamic, real-time, browser based display of agent and queue activity across different media types.
- Enable agents to treat emails like voice calls by assigning hold status to email and transferring them to other agents
- Enable supervisors to create and manage FAQ response templates.

Mitel Networks 6160 Intelligent Queue

Mitel Networks 6160 IQ is an all-in-one, scalable (supports 4-64 ports), browser based announcement solution for contact centers. It enables supervisors and managers to rapidly and intuitively:

- Provide callers in queue with pre-recorded announcements -Static Recorded Announcement
- Provide callers with their expected wait time or position in queue -TIQ Talk
- Provide callers with time of day, day of week or exception based messaging -Smart Messaging
- Perform automated attendant operations to guide callers to the most appropriate information, ACD path or extension -Smart Choice.
- Provide callers in queue with the option of entering their phone number and voice message requesting a queued call back from the contact center -Voice Callback
- Enable a customer to submit a request for contact via the web -Web Callback
- Allow a call to be routed by ANI, time of day or current queue conditions -Smart Routing

- Report on the 6160 IQ functionality -Call Detail Reporting

Mitel Networks 6170 Quality Monitoring

Mitel Networks 6170 Quality Monitoring provides contact centers with the tools required to record calls for liability purposes and/or evaluate the quality of an agents work, measure performance and improve the response of the agents.

For quality purposes Mitel Networks presents the 6170 Quality Monitoring solution, agent evaluation and selective recording software used to measure quality in the Contact Center. Used in conjunction with the 6300 call recorders – Marathon TDI solution, this product was designed from the ground up to integrate with other multi-media technologies that are becoming more and more popular in the Contact Center Environment.

Mitel Networks 6170 quality monitoring - INSPIRATION Pro features include:

- Split screen master detail hierarchy for all information at a glance
- Organized Agents and supervisors with all relevant data into organizational structure
- Easy custom forms generation for evaluating and grading quality on your criteria
- Sessions playback in full multi media integrated through browser for easy evaluation
- Full multimedia support for voice, email web-chat/ web collaboration, video, fax etc.
- Scheduler that gives resource allocation forecast and allows you to choose among criteria; random, scheduled, event or rules based
- Reporting Module that allows the user to quickly and easily generate reports as narrowly or broadly as is required. From an individual call to an aerial view of the contact center.

📝 Note: A Marathon TDI solution is required to provide the call recording for the INSPIRATION Pro. For additional details please reference the 6300 Call Recording section.

Mitel Networks 6200 Cost Management in a Contact Center

Mitel Networks 6200 Cost Management in a Contact Center provides the supervisor with all the tools they require to keep track of communications costs. It enables the supervisor to:

- Calculate call costs
- Analyze historic calling patterns for budgetary purposes
- Negotiate better phone rates
- Apply calling activity data to contact center labour scheduling and forecasting
- Maximize employee productivity and ensure resources are used appropriately
- Reduce misuse of resources by reporting on corporate email usage.

Mitel Networks 6500 Unified Communications

Mitel Networks 6500 Unified Communications deliver powerful, unique solutions. The software resides on a server, and the system responds to conversational voice commands providing a single point of access for a wide range of information.

The 6500 UC Server can support up to 30 ports of simultaneous speech, corporate directories containing up to 10,000 names, and personal directories containing up to 500 names. The 6500 UC Server connects to the 3300 ICP through the LAN using the IP protocol.

6500 Applications run on Microsoft Windows NT Server 4.0

There are three main applications:

- Mitel Networks 6500 Attendant
- Mitel Networks 6500 Unified Messaging
- Mitel Networks 6500 Enterprise Voice Portal.

There are also three purchasable options available:

- Mobility
- Calendar and Task Management
- Fax Integration.

Mitel Networks 6500 Attendant

The Mitel Networks 6500 Attendant routes incoming calls to people or departments within a company based on spoken commands. For example, users state the name of the person or department that they want to speak to, and the system routes the call to the requested party.

With this application, users can

- Place a call to any number in the company directory by stating a name, extension, or department
- Navigate through multi-level menus using voice commands
- Call into the system from their home phone or cell phone and place calls to external numbers that are programmed in the company directory, provided the users have been assigned the required system privileges
- Program their own list of frequently called numbers and then place calls to those numbers through using speech commands (registered users only)

Additional functionality includes:

- Transfer of an incoming call to another number (Mobility option)
- Interact with VoiceXML scripts (Enterprise Voice Portal option).

Mitel Networks 6500 Unified Messaging

Mitel Networks 6500 Unified Messaging is fully integrated with both Microsoft Exchange 2000 and the 6500 Attendant. Using speech recognition, it provides voice-command access to and management of all messages from a single Outlook Inbox.

With this application, users can

- Tell the system to organize and play messages based on caller, date, type, and priority, liberating users from having to view messages sequentially
- Dial calls by simply saying the contact's name (through access to Outlook's contact list)
- Check for urgent messages using the phone
- Check for messages from a particular person.

Extra functionality includes the ability to

- Manage appointments, meeting requests, or tasks using the telephone (Calendar and Task Management option)
- Forward a fax or email to another fax number using the telephone (Fax Integration Option)

Mitel Networks 6500 Enterprise Voice Portal

Mitel Networks 6500 Enterprise Voice Portal uses Voice Extensible Markup Language (VoiceXML) technology that allows interaction with the Internet and any back-end database by using voice commands. Enterprise Voice Portal enables rapid creation and modification of voice enabled web sites for enterprises. Adding the power of voice access to web sites provides access to anyone with a telephone. It can be used to power applications such as

- Voice-activated dialing
- Telephone access to corporate intranets
- Database access for queries such as parcel tracking, news, weather, and stock quotes.

During a single telephone call, users can browse through a Web-site and link to related sites using voice commands.

The 6500 Enterprise Voice Portal is available either as a stand-alone application or can work in conjunction with the 6500 Attendant or 6500 Unified Messaging.

Mitel Networks 6500 Applications Options

These applications are available as additional purchasable options on the 6500 Attendant or 6500 Unified Communications. For complete details, refer to the Mitel Networks 6500 Applications documentation.

Mobility

The Mobility option provides users with the ability to redirect the 6500 Applications calls made to their default number to one of their other programmed numbers, or to a temporary number. While Mobility is enabled, all calls made to the user's default number (by stating only the user's name)are redirected to the "Reach me at" number; however, if a caller specifically requests the user's cellr phone, pager, fax, or home phone number (for example, by stating "Bill Smith on his cell"), then the call is directed to the requested number and is not redirected.

Calendar & Task Management

Calendar & Task Management provides users with access to their Calendar and Task lists using spoken commands. Users can also review their message lists, and create, modify, or delete appointments, meetings or tasks.

Fax Integration

Fax Integration enables users to integrate with a third party fax server. this option allows users to view their faxes in the Outlook In-box, program notification, read a fax header using text-to-speech and forward their faxes to another fax machine. The faxes are stored on the fax server and the Exchange 2000 Server (a no charge option).

📝 Note: Mitel does not provide the 3rd party fax.

📝 Note: The third party fax that is supported is Right Fax

Mitel Networks 6510 Unified Messaging

Mitel Networks 6510 Unified Messaging is an advanced computer-telephony-messaging server designed to meet the voice messaging and call handling needs of small, medium, and large enterprises. The 6510 Unified Messaging platform is designed to supply a turnkey messaging solution for a single operation, or provide enhanced, hosted services for a suite of organizations integrated in one server. Providing an all-in-one communications solution, it gives users access to applications like desktop call control, unified messaging, Web messaging and administration, instant messaging, and wireless connectivity. Companies can even configure the server to emulate other messaging solutions (such as Octel) that users may already be familiar with, thus drastically reducing training time within an organization.

6510 Unified Messaging is a superior messaging system permitting ultimate customization for any organization. It is available on a Windows 2000 Server platform; supports the TCP/IP standard network protocol, and is expandable from 4 to 144 ports. Furthermore, the 6510 can be configured in different combinations. This allows both the smallest and largest companies to implement cost-effective network-telephony technology and have the tools to move toward Unified Messaging in the future.

6510 Unified Messaging compromises six major components:

Automated Attendant

- Up to 99 auto attendants can be programmed. With 6510 Unified Messaging auto attendants may program to automatically greet callers with date and time-sensitive greetings. For instance if the office is closed for the day, the auto attendant can answer with an "Office Closed" greeting, which automatically takes effect Monday to Friday from 5pm to 8am and all day Saturday and Sunday.
- The 6510 Auto Attendant can connect callers to users who are off-site by giving callers direct call access to the user's cell, home, or branch office phone number. The attendant gives callers single-digit access to departments or individuals, Fax-on-Demand options, Information mailboxes, and Interview boxes through a full-featured audio-text menu. The auto attendant also supports Intelligent Call Routing.

Multi-Lingual Support

- The 6510 offers caller services in many languages. Language-sets are currently in English, French, Spanish, Dutch, German, and Italian.

Voice Mail & Integrated Messaging

- Voice and fax messages can be delivered directly to an e-mail package such as Microsoft Exchange/Outlook™, Novell GroupWise™, Lotus Notes™, or any other SMTP and IMAP/POP3 compliant email applications being used. 6510 Unified Messaging allows integration of voice and fax messages into any mail server so you can retrieve an entire suite of message types using a single application.

- An unlimited number of Voice mail/Integrated Messaging users may use the system. These users can obtain a unified view of their email, voice mail and fax messages using their desktop email application, or their telephone by using a Text-To-Speech (TTS) option.

- The 6510 TUI enables Integrated Messaging users to configure and control their messaging options using a telephone user interface (TUI). Options that can be accessed through the TUI include checking and responding to messages, programming up to 99 greetings/greeting schedules, configuring notification schedules and call transfer options, and managing distribution lists.

Unified Messaging

- Up to 10000 unified messaging users can utilize the 6510 for complete synchronization of all their message types, complete desktop call control, and mailbox management.

- In a unified messaging configuration, a message synchronization server (MSS) or IMAP4 Gateway is setup to ensure complete synchronization of all messages among the existing message servers and the 6510. MSS uses the transport protocol MSMQ (Microsoft Message Queue), a new Microsoft technology developed to guarantee message delivery. When using MSMQ to keep MSS clients informed of changes, you ensure that all clients (servers) remains synchronized. MSMQ is a reliable transport infrastructure, providing the foundation for a reliable solution.

- Using Mitel Networks' Web Client, 6510 UM users have full mailbox control from their desktop or mobile PC through an easy-to-use, graphical user interface (GUI) that is accessible with any standard web browser. Users can change mailbox status, record or change greetings, set up notification schedules and distribution lists, listen to messages, create folders and move messages from one folder to another, all without ever picking up a phone. Desktop call control even integrates with many popular PIM applications such as Act!, Maximzer, Outlook and Goldmine, allowing UM users to identify callers at a glance and control how the call is handled.

System Administration 6510 Unified Messaging's Administrator Console is an plug-in for Microsoft Management Console (MMC). Using this application, System Administrators can access administrative tools and functionality from any LAN-based or Internet-based computer. The Administrator Console allows up to 10 System Administrators to manage the voice mail system from any location, controlling every aspect from changing a mailbox feature to adding a PBX. Mailbox configuration features under administrator control include:

- Feature Groups (Class of Service)
- Flexible Schedules
- Intelligent Dial Plan
- Wakeup Call
- Mailbox Templates
- Copy & Paste Functionality
- Customized Mailbox Layout
- Additional Mailbox Options

Optional Fax Mail and Fax Server

- Fax Mail allows users to receive faxes in their desktop mailbox. The fax board in the 6510 Unified Messaging server receives the faxes and deposits them in the appropriate mailbox(es). Fax Mail then notifies the user of the fax message. Upon opening the file, the default fax viewer displays the document. The user can then print, redirect, or save the fax. Through the mailbox, callers are prompted to press a key to print their fax or redirect it to another fax machine.

- Fax Server is an off-board (separate server) option that comprises two major components - Fax Server and Fax Client. Fax server is a powerful, high-volume document-delivery and receipt solution that reduces costs and improves efficiency. It provides real-time document delivery from the desktop to fax machines, other Fax Clients, and the Internet by controlling the sending and receiving of faxes to desktops for multiple concurrent users at once.

- Both Fax solutions support configurations of 2 to 16 analog fax ports.

Mitel Networks 9910 Commander Contact Center

The Mitel Networks 9910 Commander Contact Centre is an innovative advancement in contact centre communications and control, extending the boundaries of the customer-agent interaction to support a wide range of contact types in a completely integrated environment. Commander's patented solution provides Multimedia Interaction Management™ through the most comprehensive set of tools on the market - routing, queuing, tracking, and reporting on inbound and outbound calls, e-mail, Web Chats, Web Requests, faxes, voice mail, and blended calls (preview dialling).

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

The Commander solution transforms the contact centre into a true business interaction centre, embodying characteristics unique to the industry. Features of the system include:

- Internationalised product. The user interfaces and documentation now support localisation, including double-byte encoded languages such as Japanese.

- Complete switch independence and support. The product interfaces to any combination of switches (PBX, ACD, IP, or central office). It can utilize analog, digital, and Computer-Telephone Integration (CTI) switch link connections.

- Interoperability with external systems. Business data can be exchanged with any database or application through industry standard protocols and access methods, including ODBC, OLE, DDE, HLLAPI, and programmatic interfaces.

- Configurable methods of communication. Interactions are managed and distributed to agents using customizable prioritization and escalation strategies. Configurable alarms can be utilised to present visual and audio cues, or trigger any programmable action.

- Packaged integration. Interactions can be managed from beginning to end within the boundaries of the product through supported Interactive Voice Response (IVR) routing, queue handling, voice messaging, and voice recording functions. Alternatively, external systems can be utilised to support these functions through interoperability.

- System partitioning. Users and interactions can be displayed, managed, and tracked in logical groups through the use of queues, queue groups, and workgroups.

- Flexible levels of fault tolerance. A number of different configurations are supported allowing various levels of fault tolerance and recovery, from disk mirroring to system fail over to truly redundant constant computing.

- Seamless, comprehensive, transactional data collection. Volumes of contact centre data are collected and shared, providing insightful knowledge that supports best business practices at administrative, operational, and management levels. Report scheduling, Web publishing, and a fully supported interface to Seagate Crystal Reports™ allows complete customization of content and presentation to address business needs.

- Access to archived interactions. The Interaction Vault allows the contact centre managers and agents to search and query the Decision Management System database with a variety of historical searches that include the viewing of all types of interactions, and all of their associated detail (i.e., date and time of interaction, agent taking part in interaction, type of interaction, etc.).

- Customizable Interaction Prioritization. Allows the organization to use business rules to determine the prioritization of interaction handling.

- Interaction Queue Management. Queued Interaction Manager (QIM) is a Web-based utility provides an efficient method to manage queue space by allowing multiple deletions of unwanted interactions such as e-mail, voice mail, and call-backs from the queue.

Note: This solution is available in the UK only.

Desktop Applications

Mitel Networks 5810 PDA Application

The Mitel Networks 5810 PDA applications allow a user with a Palm® Handheld™ Personal Digital Assistant (PDA) to dial numbers and activate features on a Mitel Networks 5020 or 5220 IP phone that has a Mitel Networks IrDA Module, a 5140 IP Appliance, or a 5240 IP Appliance.

- The Telephony Features Interface (TFI) application allows the creation of "soft" buttons on a PDA that, when activated, emulate the buttons on the users IP phone.
- The Dial by Address Book (DBAB) application enhances the PDA's existing Address Book function by allowing the user to dial numbers from their address list on the PDA including the ability to create or modify a dialing prefix.

Note: The Mitel Networks IrDA Module, which is available as an add-on module for the Mitel Networks 5020 IP Phone, the Mitel Networks 5220 IP Phone and is integral to the Mitel Networks 5140 and 5240 IP Appliance, provide the connectivity between the telephone and the PDA.

Note: The 5810 PDA application is only available in English.

Mitel Networks 6600 Your Assistant

Mitel Networks 6600 Your Assistant enables customers to effectively manage telephony and data communication with features that enhance productivity such as PC-based call handling, visual conference call management, call annotation timing and recording and integrated MSN messaging. 6600 Your Assistant is available in two different formats 6600 Your Assistant and 6600 Your Assistant Pro.

6600 Your Assistant features include:

- Visual Conference Call Management
- Call Forward profiles
- Caller line ID Screen Pop with PIM integration
- Call Timer and annotation
- Call history
- Instant Messaging integration
- Favourites/Quick list
- Do not disturb
- Configurable interface

6600 Your Assistant Pro includes of the features listed above and provides these additional features:

- Softphone
- Record calls

Manual Maker

Manual Maker is a tool for creating customized user guides. This tool provides the ability to tailor a user guide to specified systems, phone models, programmed features, feature activation methods and/or attendant consoles (voice mail, ACD, and subattendant features).

Manual Maker is available to Mitel Networks authorized partners via Mitel Online.

Peripherals

The following peripheral devices are supported on the 3300 ICP.

- Mitel Networks 5001 IP Phone
- Mitel Networks 5005 IP Phone
- Mitel Networks 5010 IP Phone
- Mitel Networks 5020 IP Phone
- Mitel Networks 5140 IP Phone
- Mitel Networks 5201 IP Phone
- Mitel Networks 5205 IP Phone
- Mitel Networks 5215 IP Phone
- Mitel Networks 5220 IP Phone
- Mitel Networks 5230 IP Appliance
- Mitel Networks 5240 IP Appliance
- Mitel Networks 5303 Conference Phone
- Mitel Networks 5305 IP Office Conference Unit
- Mitel Networks 5310 IP Board Room Conference Unit
- Mitel Networks 5550 IP Console.

The following accessories are supported:

- Mitel Networks 5410 Programmable Key Module 12
- Mitel Networks 5415 Programmable Key Module 48
- Mitel Networks 5412 IP Programmable Key Module
- Mitel Networks 5448 IP Programmable Key Module
- Mitel Networks 5423 IrDA Module
- Mitel Networks 5424 IrDA Module
- Mitel Networks 5485 IP Paging Unit
- Mitel Networks 3300 Power Dongle (cisco compliant)
- PowerDsine 24PT In-line Power Unit.

The following Symbol devices are supported:

- Symbol MiNET Wireless Phone
- Spectrum24 Access Points.

The following legacy telephones and consoles, which are available for purchase, can be used with the 3300 ICP if a peripheral unit with a DNI card is connected to the system:

- SUPERSET 4001 Telephone
- SUPERSET 4015 Telephone
- SUPERSET 4025 Telephone
- SUPERSET 4150 Telephone
- SUPERCONSOLE 1000

Analog telephones can also be connected to a 3300 ICP using a 3300 Universal ASU, 3300 ASU. or peripheral unit with an ONS Line Card or LS/GS trunk card installed.

Desktop

Mitel Networks 5001 IP Phone

The Mitel Networks 5001 IP Phone is a low-cost, single port entry-level IP telephone that connects to a 10BaseT Ethernet network. Features of the telephone include:

- Three fixed-function keys: Hold, Message, and Transfer/Conference
- Handset and Ringer Control
- Message Waiting Lamp
- Wall-mounting
- Multiple powering options.

LL0067

Mitel Networks 5005 IP Phone

The Mitel Networks 5005 IP Phone is a low-cost, single port multiline IP telephone that connects to a 10BaseT Ethernet network. Features of the telephone include:

- One line by 20-character backlit alpha-numeric liquid crystal display (LCD) with contrast adjustment
- 20 programmable feature keys (six pre-assigned)
- Two fixed-function keys: Hold, Cancel
- Handset and Ringer volume Control
- Message Waiting Lamp
- Wall-mounting
- Multiple powering options.

LL0066

Mitel Networks 5010 IP Phone

The Mitel Networks 5010 IP Phone is a dual port, multiline IP telephone that connects directly to a 10/100BaseT Ethernet network. Features of the telephone include

- Two line by 20-character alpha-numeric backlit LCD with contrast adjustments
- 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
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide LAN connectivity for both your phone and PC
- Dedicated headset port
- Multiple powering options.

LL0029

Mitel Networks 5020 IP Phone

The Mitel Networks 5020 IP Phone is dual-port multiline IP speakerphone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Two line by 20-character alpha-numeric backlit display with contrast control
- Three context sensitive softkeys for feature access
- 14 programmable feature/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
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide LAN for both your phone and PC
- Dedicated headset port
- Multiple powering options
- Module support (IrDA, PKMs, Conference Units).

LL0030

Mitel Networks 5140 IP Appliance

The Mitel Networks 5140 IP Appliance is an exciting advance in IP telephony. It combines proven, high-quality communications with superior feature access for easy information retrieval. This device adds IP value to the desktop by with improved integration of desktop devices and software server applications. It also provides portal access to the LAN network. Features of this device include

- Audio-device controls
- Large LCD screen with backlighting for easy viewing (320 x 240 VGA Display)
- Integrated Infrared transceiver lens (IrDA) for use with Mitel Networks 5810 PDA Application
- Six direct access command keys that perform context-sensitive actions shown on the LCD display
- Nine programmable feature keys for interacting with items shown on the LCD display
- Four navigation keys enabling you to scroll and move around pages on the display screen
- Three telephony feature keys
- Eight keys for easy access to 5140 IP Appliance applications
- Ringing Indicator
- Message Waiting lamp
- Two 10/100BaseT Ethernet ports
- Embedded Help
- Dedicated headset jack
- Multiple powering options.

The 5140 IP Appliance has its own applications which are hosted on a server and accessed through an integrated browser. Applications of the 5140 IP Appliance include

- Corporate Directory allows users to scroll through names and search by last name, full name, department, and location
- Personal Directory allows users to maintain a list of personal contacts
- Visual Voice Mail allows users to view the contents of their voice mailbox on the LCD
- Call Log records the last 60 incoming and outgoing calls (detailing caller ID, duration, date, and time) made to and from the 5140 IP Appliance.
- HTML Browsing with personal bookmarks to allow quick access to a customized list of URLs
- Services allows the administrator to program corporate URLs that can be accessed easily through the 5140 IP Appliance

For detailed information, refer to the Mitel Networks 5140 IP Appliance help system.

LL0020

Mitel Networks 5201 IP Phone

The Mitel Networks 5201 IP Phone is a low-cost, single port entry-level IP telephone that connects to a 10BaseT Ethernet network. Features of the newly designed telephone include:

- Three fixed-function keys: Hold, Message, and Transfer/Conference
- Handset and Ringer volume Control
- Message Waiting Lamp
- Wall-mounting

NN0009

Mitel Networks 5205 IP Phone

The Mitel Networks 5205 IP Phone is a low-cost IP telephone that connects to a 10BaseT Ethernet network. Features of the newly designed telephone include:

- 20-character alpha-numeric liquid crystal display (LCD)
- 20 feature keys (6 pre-assigned)
- Two fixed-function keys: Hold, Cancel
- Handset and Ringer Control
- Message Waiting Lamp
- Wall-mounting

NN0010

Mitel Networks 5215 IP Phone

The Mitel Networks 5215 IP Phone is a dual port, multiline IP speakerphone that connects directly to a 10/100BaseT Ethernet network. Features of the newly designed telephone include

- Two line by 20-character alpha-numeric backlit liquid crystal display (LCD)
- Seven programmable feature/line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Speaker, and Microphone.
- Handsfree operation (half-duplex)
- Automatic selection of prime line or ringing line
- Key selection of non-prime line
- Handset and ringer volume controls
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated Headset port
- Multiple powering options.

NN0011

Mitel Networks 5220 IP Phone

The Mitel Networks 5220 IP Phone is an IP telephone that connects to a 10/100BaseT Ethernet network. Features of the newly designed telephone include

- Two line by 20-character alpha-numeric LCD with contrast control
- Three softkeys for feature access
- 14 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
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated headset port
- Multiple powering options

NN0012

47

Mitel Networks 5230 IP Appliance

The Mitel Networks 5230 IP Appliance combines the power of a desktop telephone with a Personal Digital Assistant (PDA). A software CD is supplied with the phone that enables the PDA to communicate with the Appliance and use applications such as Dial By Address Book which enables users to dial a number from the contact list of their PDA and the viewing of call logs.

The phone enables customers to take advantage of the full suite of features that the 3300 ICP provides either through the PDA's Graphical User Interface through the integrated LCD screen.

Hardware features include:

- 2 X 20-character alpha-numeric backlit liquid crystal display (LCD) with contrast adjustments
- Six fixed-function keys: Microphone Mute, Speaker Engage, Speaker Volume Control, Hold, and Transfer/Conference
- Three context sensitive softkeys for feature access
- PDA Interface Module/Personality Module provides support for the following PDAs Compaq iPAQ H3760 and IPAQ H3850. This cradle provides both the charging capability and the hot synching of the docked PDA.
- Handsfree operation (half duplex)
- Integrated serial/USB port via the PDA Interface Module/Personality Module.
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Supports Line Powering of the telephone (this excludes PDA charging)
- Jack to connect the PDA charging cable
- Handset
- Dedicated Headset Port
- Mounted stylus holder
- External host USB connection

5230 IP Appliance with PDA

5230 IP Appliance without PDA

Mitel Networks 5240 IP Appliance

The Mitel Networks 5240 IP Appliance combines proven, high-quality communications with superior feature access for easy information retrieval. This device adds IP value to the desktop by offering improved integration with desktop devices and software server applications. It also provides portal access to the LAN network. Features of this device include

- Audio-device controls
- Large LCD screen with backlighting for easy viewing (320 x 240 VGA Display)
- Integrated Infrared transceiver lens (IrDA) for use with Mitel Networks 5810 PDA Application
- Six direct access command keys that perform context-sensitive actions shown on the LCD display
- Nine programmable feature keys for interacting with items shown on the LCD display
- Four navigation keys enabling you to scroll and move around pages on the display screen
- Three telephony feature keys
- Eight keys for easy access to 5140 IP Appliance applications
- Ringing Indicator
- Message Waiting lamp
- Two 10/100BaseT Ethernet ports
- Embedded Help
- Dedicated headset jack
- Multiple powering options.

The 5240 IP Appliance has its own applications that are hosted on a server and accessed through an integrated browser. All applications are available in the 10 languages supported by the 3330 ICP. Applications of the 5240 IP Appliance include

- Corporate Directory allows users to scroll through names and search by last name, full name, department, and location
- Personal Directory allows users to maintain a list of personal contacts
- Visual Voice Mail allows users to view the contents of their voice mailbox on the LCD
- Call Log records the last 60 incoming and outgoing calls (detailing caller ID, duration, date, and time) made to and from the 5140 IP Appliance.
- HTML Browsing with personal bookmarks to allow quick access to a customized list of URLs
- Services allows the administrator to program corporate URLs that can be accessed easily through the 5140 IP Appliance

NN0057

Mitel Networks 5303 Conference Phone

The Mitel Networks 5303 Conference Phone is a stand-alone, analog telephone conferencing unit. The product consists of a "saucer" and a "tablet" connected by a short cable. The saucer encompasses the speakers/ microphones while the tablet provides the dial pad, softkeys, volume controls, mute and backlit display.

The Mitel Networks 5303 conference Phone builds on Mitel Networks' tremendous expertise in acoustic design. It is the only analog conference phone on the market with exclusive acoustic beamforming technology that automatically focuses the microphone on individual speakers to eliminate annoying background noise and side conversations.

The 5303 Conference Phone quickly and easily plugs into any telephone wall jack or standard office telephone system that connects to any Loop Start (LS) or ON premiSe (ONS) line on the 3300 ICP.

Features of this unit include

- Full Duplex audio
- Programmable softkeys that support Speed Calls, Redial, Flash, Language Selection, Transfer/Conference
- Acoustic beam- forming that controls near end, far end and double-talk, and locates direction of speech
- Noise Reduction and automatic gain control to eliminate background noise
- Dynamic allocation of microphones to activate speakers
- 40 character 2 line LCD display with backlighting and contrast control
- Calling Line ID
- Speed Dial programming (up to nine stored numbers)
- Nine pre-programmed speed dials
- Time of day and date programming
- Echo cancellation
- 12 key alpha numeric keypad
- Volume up/down controls
- Microphone Mute key
- Presentation Mode Key (locks microphone to current speaker)
- Length of call feature
- Available in two colors: Silver Ghost and Night Sky

LL0046

Mitel Networks 5305 IP Office Conference Unit

The Mitel Networks 5305 IP Office Conference Unit uses acoustic beam-forming technology to ensure superior audio performance. It connects to the headset port of the 5020 IP Phone or 5220 IP Phone and is designed for optimal use in any 10 feet by 10 feet (3 meters by 3 meters) private office.

Features of the conference unit include:

- Full duplex operation
- Acoustic beam-forming technology that controls near end, far end, and double talk, and locates direction of speech
- Noise reduction and automatic gain control to eliminate background noise
- High fidelity speaker
- Power Supply from a 24VDC wall adapter
- Simple installation
- Side Control Unit with mute, up/down volume controls and on/off key
- Available in Night Sky (dark grey) colour only.

The 5305 IP Conference Unit package includes a speaker unit and a side control unit. An optional mouse controller is available.

LL0073

Mitel Networks 5310 IP Board Room Conference Unit

The Mitel Networks 5310 IP Board Room Conference Unit is a full duplex, high quality conference unit that uses acoustic beam-forming technology for superior performance. It connects to the headset

port of 5020 IP Phone or 5220 IP Phone and is designed for optimal performance in any 15 feet by 25 feet (4.5 meters by 7.6 meters) boardroom.

Features of the conference unit include

- Full duplex audio
- Acoustic beam-forming technology that controls near end, far end, and double talk, and also locates direction of speech
- Noise reduction and automatic gain control to eliminate background noise
- High fidelity speaker
- Presentation mode that locks the microphone array to the current speaker.
- Seven dual color LEDs for visual confirmation that the unit has picked up the speaker's voice
- Power supply from a 24VDC wall adapter
- Simple installation
- Side control unit with mute key, up/down volume controls, presentation mode key and on/off key.
- Available in two colors: Silver Ghost and Night Sky

The 5310 IP Conference Unit package includes a 5020 IP Phone, a speaker unit, and a side control unit. An optional mouse controller is available.

LL0058

SUPERSET 4001

The SUPERSET 4001 connects to a DNI card in the Peripheral unit. Providing access to basic system functionality. It 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
- Ringer pitch control
- Message Waiting lamp

ES0007

SUPERSET 4015

The SUPERSET 4015 connects to a DNI card in the Peripheral unit. It is a multiline, digital telephone with

- 20-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
- Ringer pitch control
- Message Waiting lamp

ES0008

SUPERSET 4025

The SUPERSET 4025 connects to a DNI card in the Peripheral unit. It is a multiline, digital telephone with

- 20-character alpha-numeric liquid crystal display (LCD) with contrast control
- Three softkeys for feature access
- 14 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
- Ringer pitch control
- Message Waiting lamp

The SUPERSET 4025 supports PKM Interface Modules for connection to additional devices.

ES0009

SUPERSET 4150

The SUPERSET 4150 telephone connects to a DNI card in the Peripheral unit. It is a multiline, digital telephone with

- 40-character alpha-numeric liquid crystal display (LCD) with contrast control and six touch-sensitive softkey areas for feature access
- 14 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
- Ringer pitch control
- Message Waiting lamp

The SUPERSET 4150 supports PKM Interface Modules that let you connect to additional devices.

ES0011

Symbol MiNET Wireless Phone

The Symbol MiNET Wireless Phone is a small, lightweight, wireless telephone designed to operate over Spectrum24 wireless data networks by using Voice-over-IP. This phone is available in Direct sequencing only and can only be used with an 11MB Access Point (802.11b).

This phone and supporting apparatus provide greater flexibility to the 3300 ICP. The phone interfaces directly with the controller which provides full system functionality. The Symbol MiNET Wireless Phone acts like a 5020 IP or 5220 IP Phone with the exception of multiline keys, multiple speed dial buttons or anything that requires speakerphone functionality such as Group Page.

Features of the phone include

- Three line by twelve character backlit display
- Headset port
- Programmable settings for Ring Type, Ring Tone, key Volume, Backlight, and Contrast Control
- Vibrating, visual and audio ring signals
- Message Waiting Indicator
- Signal and battery strength indicator
- Shared and personal user mode support
- DHCP Support
- Function key (gives access to phone menu through softkeys)
- Swap Key (enables user to swap between phone mode and user mode)
- Serial port (to download software into the phone)
- Out of Range warning notification
- Supports G.711 and G.729 compression
- 1.5 hours talk time, 30 hours standby
- Signaling power 100mW
- Access to the complete range of telephony features available on 3300 ICP including speed dials, transfer/conference, call waiting, call forwarding, call hold do not disturb, and ACD agent functionality. For the complete list of system features refer to the 3300 ICP Features table.

Embedded wireless communication features include

- Dynamic rate scaling
- Symbol Wireless voice Prioritization on Spectrum24 High Rate Wireless LAN
- Pre-emptive Roaming for optimum voice quality and bandwidth maximization
- 40 and 128 bit WEP encryption security
- IEEE 802.11d International Roaming.

EE0535

Consoles

SUPERCONSOLE 1000

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 four-line by 80-character alphanumeric display shows source and destination, time and date, call waiting, and station information (such as COS and COR values). Macros can be programmed to facilitate the transfer of calls to voice mail, recover calls released to the wrong extension, and dial frequently called numbers using one button.

The SUPERCONSOLE 1000 connects to a DNI card in the Peripheral unit.

The console has

- 14 hardkeys
- Four programmable firmkeys -for access to purchased options such as Hotel/Motel
- 10 softkeys
- A dial pad (for both alphabetic and numeric input)
- Backlit display
- Volume controls
- Integral handset
- Headset connector
- An RS-232 serial printer port.
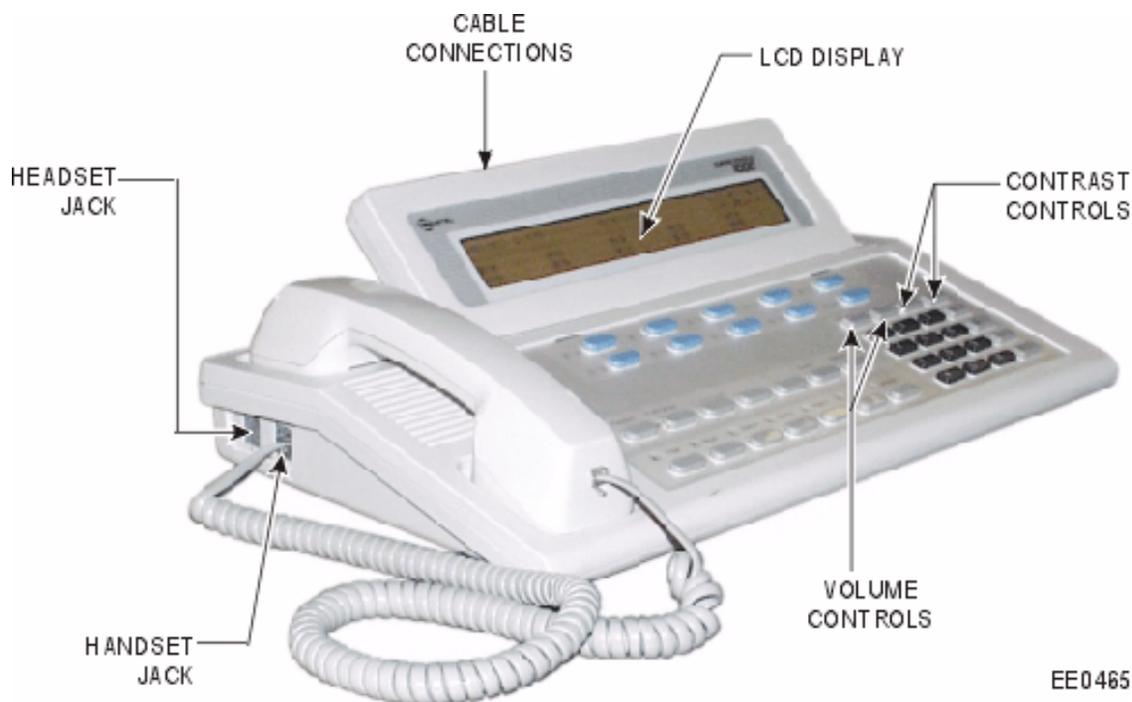
Mitel Networks 5550 IP Console

The Mitel Networks 5550 IP Console is an advanced PC-based console and administration application. It has a highly intuitive Graphical User Interface (GUI) including screen based call status and call handling prompts. A telephony keypad and dual handset/headset jack provide fast, efficient attendant call handling on the Mitel Networks 3300 ICP.

This 5550 IP Console is ideal for both departmental and enterprise attendants requiring fast and easy access to call-processing functionality, and the ability to use other applications in the off-peak call volume hours.

Some of the console features include

- Specialized telephony keypad for dialing, call processing, and access to 3300 ICP features and applications
- Highly intuitive Graphical User Interface (GUI)
- One-button access to programmable key functions
- On-screen scratch pad window for note taking and messaging, and storage of speed dial numbers
- On-screen bulletin board for sharing information with other 5550 IP Console attendants on the system
- Retrieve key to retrieve calls forwarded to the wrong extension
- Single key to transfer calls to voice mail
- Language Support for English, French, Spanish, German, Dutch, Italian, and Portuguese
- Console telephony hardware is available in two colours: light grey and dark grey.

The 5550 IP Console solution consists of the telephony hardware and the console application software that will enable it to run on a customer-supplied PC. This PC should have as a minimum: 450MHz or faster Pentium-compatible processor, Microsoft Windows 2000 Professional, Microsoft Windows 98 or Microsoft Windows Millennium, 128MB of available RAM, 4GB hard drive, 17-inch SVGA monitor, CD-ROM drive, AT 101 enhanced keyboard, mouse, and a VLAN-aware Network Interface Card (NIC).

The package includes the following items:

- Telephony Keypad
- 24VDC power adapter
- Mitel Networks 5000 series handset and cord
- Handset Cradle
- Ethernet cable 10/100BaseT
- CD-ROM containing the console application software
- Designation labels for the programmable keys
- Quick Start Guide
- Installation Guide.
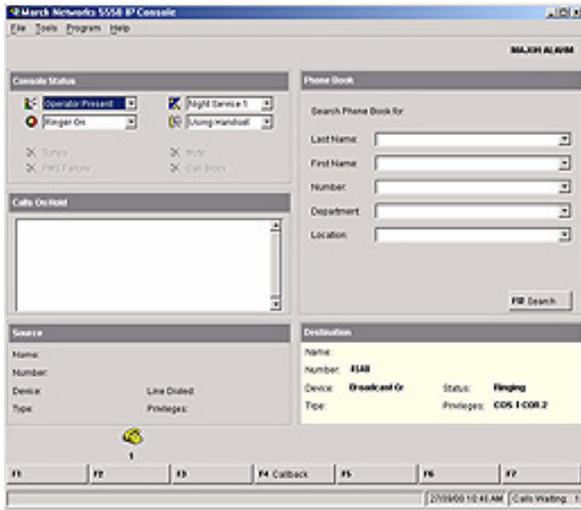
EE0527

Accessories

Mitel Networks 5410 Programmable Key Module

The Mitel Networks 5410 Programmable Key Module provides 12 additional personal keys for a 5020 IP Phone. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The 5410 PKM unit connects to a 5020 IP Phone through a Mitel Networks 5421 PKM Interface Module (IM). which is installed separately at the base of the telephone and is only compatible with 5020 IP Phones.

ES0124

Mitel Networks 5415 Programmable Key Module

The Mitel Networks 5415 Programmable Key Module 48 provides additional feature keys for a 5020 IP Phone. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The 5415 PKM unit connects to a 5020 IP Phone through a Mitel Networks 5421 PKM Interface Module (IM) which is installed separately at the base of the telephone. A second 5415 PKM can connect to the first to provide 48 additional feature keys.

LL0035

Mitel Networks 5412 IP Programmable Key Module

The Mitel Networks 5412 IP Programmable Key Module provides 12 additional personal keys for a 5220 IP Phone. They can be programmed as feature keys, speed call keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The 5412 PKM unit connects to a 5220 IP Phone by using a Mitel Networks 5422 PKM Interface Module (IM). The PKM IM is installed separately at the base of telephone and is only compatible with 5220 IP Phones.

5220 IP Phone with 5412 PKM

Mitel Networks 5448 IP Programmable Key Module

The Mitel Networks 5448 Programmable Key Module provides 48 additional feature keys for a 5220 IP Phone. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or line appearance keys. Each key has a Line Status Indicator that works the same way as those on the associated telephone. The keys can be programmed through the telephone.

The 5448 PKM unit connects to a 5220 IP Phone by using a Mitel Networks 5422 PKM Interface Module (IM). A second 5448 PKM can connect to the first to provide for 48 additional feature keys. The 5422 PKM IM is installed separately at the base of telephone and is only compatible with 5220 IP Phones.

NN0015

Mitel Networks 5423 IrDA Module

The Mitel Networks 5423 IrDA (Infrared Display Adapter) Module attaches to a 5020 IP Phone. Mitel Networks 5810 PDA application software must also be installed on the Palm Personal Digital Assistant (PDA) operating system. A wireless connection between the telephone and the Palm PDA is established through the infrared ports.

This optional module gives the user the ability to dial telephone numbers directly from the Palm PDA. If users find themselves away from their desks, they can point their Palm PDA at any telephone with the attached module, and access features and telephone numbers programmed at their own extension number.

LL0023

Mitel Networks 5424 IrDA Module

The Mitel Networks 5424 IrDA (Infrared Display Adapter) Module attaches to a 5220 IP Phone. Mitel Networks 5810 PDA application software must also be installed on the Palm Personal Digital Assistant (PDA) operating system. A wireless connection between the telephone and the Palm PDA is established through the infrared ports.

This optional module gives the user the ability to dial telephone numbers directly from the Palm PDA. If users find themselves away from their desks, they can point their Palm PDA at any telephone with the attached module, and access features and telephone numbers programmed at their own extension number.

5220 IP Phone with 5424 IrDA Module

Mitel Networks 5485 IP Paging Unit

The Mitel Networks 5485 IP Paging Unit is an optional module that provides paging functionality on the 3300 ICP.

The IP Paging Unit is installed as a stand-alone or a wall-mounted unit. Two LEDs provide basic status information. The unit connects to the LAN using an RJ-45 cable and is powered by a 24 VDC power adapter which is supplied.

Each IP Paging Unit supports one paging zone.

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

For detailed information, refer to the documentation shipped with the product.

Mitel Networks 3300 Power Dongle (cisco compliant)

The Mitel Networks 3300 Power Dongle enables Mitel Networks IP telephones to be powered by Cisco 600*/4006, 3524 series of catalyst switches or the Cisco mid-span power hub.

IP0448

PowerDsine 24PT In-line Power Unit

PowerDsine 24PT In-line Power Unit supplies uninterrupted power to Mitel Networks IP Phones and Symbol Spectrum Access Points. By using this unit to power IP devices customers can reduce the cost of installing separate power cabling to each endpoint.

Each unit can be used to provide remote power feeding for 24 IP telephones from a centralized universal 100/240VAC, 50/60Hz AC input.

The PowerDsine 24PT In-line Power Unit is connected in series to an Ethernet switch. The data output jack on the switch connects to the input jack on the PowerDsine 24PT In-line Power Unit , and the data/power output jack on the PowerDsine 24PT In-line Power Unit connects to the data input jack on the IP telephone.

Features of this unit include:

- IEEE 802.3af compliant power outputs
- Independent overload & short circuit protection for each channel
- 24 10/100 BaseT data and power combined channels
- Universal 100-240VAC 50/60Hz power unit
- Standard 19" 1U rack mounting hardware
- Bicolor LED port status indicators
- Rs-232 serial port for software download and monitoring.

IP0430

Symbol Spectrum24 Access Point

The Spectrum24 Ethernet Access Points (AP's), manufactured by Symbol Technologies, function as a Media Access Control (MAC) bridge between wired LANs and wireless endpoints (NetVision telephony devices).

The Spectrum24 AP-4100 Series Access Points use the Direct Sequence Spread Spectrum used for in-building wireless networks. This Access Point operates at up to 11 Mbps and conforms to the IEEE 802.11(b) standard.

Features of the AP-4100 series Access Points include:

- Built-in diagnostic capabilities with power up self check
- Wireless MAC interface
- 10/100 BaseT Ethernet port interface with full-speed filtering
- PC/AT Serial Port Interface
- Data encryption
- Simple Network Management Protocol
- Support for roaming across routers
- Support for up to 127 mobile units
- DHCP support
- DNS support.

For more information on Symbol products visit Symbol Technologies website at www.symbol.com.

Network

Lines

The 3300 ICP supports the following types of internal voice lines:

- A 10/100BaseT Ethernet connection is required for Mitel Networks IP telephones to connect through an Ethernet LAN to a 3300 Controller. These lines are supported by the 3300 controller.

- On-Premises (ONS) lines (24V per port) are for industry-standard DTMF or rotary dial telephones. The external loop resistance on an ONS line must be 600 ohms or less, and the loop length must be 5000 ft (1500m) or less on 26-gauge wire. These lines are supported by the 3300 Universal ASU, the 3300 ASU and the ONS line card in the Peripheral unit or MICRO LIGHT. Note: Rotary dial telephones are not supported by the 3300 Universal ASU or the 3300 ASU.

- Off-Premises (OPS) lines (48V per port) are for industry-standard telephones where the external loop resistance exceeds 600 ohms or where lightning surge protection is required. The maximum resistance on an OPS line must be 1800 ohms or less, and the loop length must be 19,000 feet (5800 meters) or less on 26-gauge wire. These lines are supported by the OPS line card and are only available if a Peripheral unit or MICRO LIGHT is installed.

- Digital Network Interface (DNI) lines provide an interface for Mitel Networks digital telephones, and consoles. The maximum loop resistance on a DNI line must be 280 ohms or less, and the loop length must be 3300 feet (1000 meters) or less on 26-guage wire. These lines are supported by the DNI Line card and are only available if a Peripheral unit or MICRO LIGHT is installed.

Trunks

The system can connect to the Public Switched Telephone Network (PSTN) or to private networks over both digital and analog trunks.

The following digital links are supported:

- DS1 Links -The system supports D4, Q.Sig, MSDN/DPNSS, Primary Rate ISDN (DM-250, DMS-100, Bellcore National ISDN, 4ESS, NI-2, 5ESS NI2, NI13), and XNET over PRI protocols. The system connects to DS1 links through the 3300 Universal NSU or the DS1 Formatter card in the DSU cabinet or MICRO LIGHT.

- E1 Links -The system supports DASS II, MSDN/DPNSS, Q.Sig, Primary Rate ISDN (Euro ISDN (CTR4)), and XNET over PRI protocols. The system connects to E1 links through the 3300 Universal NSU or the CEPT Formatter card in the DSU cabinet or MICRO LIGHT.

- R2 Links -The system supports the CCITT Blue Book, Volume IV, Fascicle VI.4, Specifications of the Signaling System R2, Recommendations Q.440 to Q.490 (with the exception of Echo Suppression (Q.479), Test Calls (Q.490) and international signals). The system connects to R2 links through the 3300 R2 NSU.

- PRI Links -The system supports DM-250, DMS-100, Bellcore National ISDN, 4ESS, NI-2, 5ESS NI2, NI13, and Euro ISDN (CTR4) protocols. The system connects to PRI links through the 3300 Universal NSU or the PRI card in the DSU cabinet or MICRO LIGHT.

- BRI Links -The system supports Euro ISDN 2B + D, Basic Rate Interface, or the North American ISDN-1 and ISDN-2 protocols. The system connects to BRI links using a DPNSS link from a Universal NSU or CEPT Formatter card through the 3300 BRI NSU or the BRI card in the DSU cabinet or MICRO LIGHT.

The following analog trunks are supported:

- Analog CO trunks- These trunks use the loop start (LS) ports on the 3300 Universal ASU or the loop start/ground start (LS/GS) card in the peripheral cabinet to interface with the system.

- E&M trunks - these trunks use the E&M trunk card in the peripheral cabinet to interface with the system. The card can be configured for either 2-wire or 4-wire operation. Type 1 through Type V circuits are supported.

- Direct Inward Dial and Tie trunks -These trunks use the DID/Loop Tie trunk card in the peripheral cabinet to interface with the system.

IP Networking

IP Networking provides an integrated networking solution that allows voice and signaling data to be transported over the existing LAN/WAN infrastructure between multiple 3300 ICPs. MSDN/DPNSS features are supported over IP Networking.

Each 3300 ICP supports up to 2000 IP trunks allowing a 'cluster' of up to eighty 3300 ICP systems to work as a single integrated voice system. Each pair of 3300 ICP systems can be connected by up to 200 IP trunks.

Note: A Mitel Networks 3800 IP Trunking Gateway adjuncts to a SX-2000 or 3200 ICP cannot be directly connected to a 3300 ICP. To support this configuration the 3300 ICP would also require an adjunct 3800 IP Gateway.

Compression

The 3300 ICP provides the option of G.729 voice compression. Licenses and additional 3300 DSP modules can be purchased to enable this feature to a maximum 64 compression channels. This feature applies compression to a call whose voice path traverses the TDM/analog to IP fabric or vise-versa, including calls made between TDM/analog devices when call setup is established over an IP network.

IP phone to IP phone calls also support compression. Compression for this scenario is applied by DSP resources in the phones and does not require compression licenses.

The compression of a standard call effectively reduces the bandwidth required per call from 64kbps to approximately 8kbps plus packet overhead.

ISDN (Integrated Services Digital Network)

ISDN Support

The Integrated Services Digital Network (ISDN) transmits voice, data, and video at high speeds. ISDN services can be deployed and accessed at enterprise, department, and desktop levels by adding the 3300 Universal NSU with PRI or the 3300 BRI NSU to the 3300 ICP.

LAN traffic can also be carried over existing private or public digital network connections on Euro ISDN, DASSII (public access) protocols, or even on a private MSDN/DPNSS network using ISDN connections to a router.

ISDN Connectivity

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 3300 ICP supports multiple ISDN protocols and provides ISDN connectivity. The system connects with the ISDN public network and data devices (such as routers, video conferencing equipment, and servers) by using Primary Rate Interface (PRI) or Basic Rate Interface (BRI). ISDN takes advantage of the following 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
- Trunk Diagnostics
- NFAS (Non-Facility Associated Signaling)
- Remote LAN Access.

ISDN Primary Rate Interface

ISDN Primary Rate Interface (PRI) has become 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 conferences, and a host of other inbound and outbound data applications.

Using ISDN PRI 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 trunks can be reduced by 10 to 15 percent. On outbound calls, the system requests the required service from the Network and the trunk takes on the requested characteristics for the duration of the call.

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

R2

The 3300 R2 NSU provides access to the R2 National Public Switched Telephone Network (PSTN) with MF-R2 digital trunk signaling. The 3300 R2 NSU supports the CCITT Blue Book, Volume VI, Fascicle VI.4, Specifications of Signaling System R2, Recommendations Q.440 to Q.490 (with the exception of Q.479 Echo Suppression, Q.490 Test Calls and international signals).

Many countries use R2 signaling but do not adhere to the CCITT recommendations in their entirety. The 3300 ICP is completely flexible and supports regional variations of the R2 protocol. Line signaling, tone interpretation, and timing parameters for the converter can be adapted to suit any national or regional requirement. For example

- Line signaling features allow you to program up to four bits to define the incoming and outgoing patterns for line signals such as Idle and Answer
- Register signaling features allow you to program the type of address signaling termination (signaled or timed) and whether signaling should be fully-compelled or semi-compelled. These features allow the individual definition of each register signaling tone.

Traffic and Performance Specifications

Traffic and Performance

Criteria	Result	
Busy Hour Call Completions (BHCC)*	5471 per hour	
Response Time Specification	**Internal Test Limit**	**Networked Test Limit (2 nodes)**
	Delay to Dial Tone P.99 < 1000 ms Dial Tone Cut Off Delay P.99 < 500 ms Post-Dialing Delay P.99 < 1500 ms Connecting Delay P.99 < 400 ms	Delay to Dial Tone P.99 < 1000 ms Dial Tone Cut Off Delay P.99 < 500 ms Post-Dialing Delay P.99 < 2070 ms Connecting Delay P.99 < 1180 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.		

Trunking

Configuration	Trunks Echo Channels	Calls per hour From IP sets	Erlangs E2T Resource	CCS E2T Resource
ACD 50 Ports	3 Trunks (T1)	1350	37.5	1350
			64 Echo Channels	
ACD 100 Ports	4 Trunks (T1)	2700	74.4	2678
			128 Echo Channels	
700 IP	4 Trunks (T1)	3521 (total 4200 CCS)	75.2	2707
			128 Echo Channels	
604 IP and 96 ONS	4 Trunks (T1)	3521 (total 4200 CCS)	70.0	2520
			128 Echo Channels	
250 IP	2 Trunks (T1)	1307 (total 1500 CCS)	26.2	966

Attendant Console Specifications

Number of Operators required (at different system traffic rates) against number of lines									
Lines	System Traffic 4CCS (Low)			System Traffic 6CCS (Medium)			System Traffic 12CCS (High)		
	25%	55% (typical)	85%	25%	55% (typical)	85%	25%	55% (typical)	85%
100	1	2	2	2	2	2	2	3	3
200	2	2	3	2	3	3	3	4	5
300	2	3	3	2	3	4	3	5	7
400	2	3	4	3	4	5	4	6	8
500	2	3	4	3	4	6	4*	7*	10*
600	3	4	5	3	5	7	5*	8*	12*
700	3	4	6	4	6	8	6*	10*	13*
*Note: System blocking will reduce the number of lines that can be handled at this traffic rate so the number of operators will be lower.									

The following assumptions apply to the table above:

- Majority of calls handled by the operator are for incoming trunk traffic.

- Calls are answered on average within 10 seconds.

- Calls are handled (transferred/dropped) within an average of 20 seconds.

- 85% of all calls are handled within these time limits.

- Table shows recommended number of operators. Local traffic conditions may increase or decrease these values, and hold times may be higher requiring more operators (or vice versa).

- Table shows the quantity of incoming trunk calls handled by operator as 25%, 55%, and 85% of all incoming calls. Remaining calls are handled through direct connection, diversion to voice mail, or IVR equipment (IVR can also be considered as an operator replacement).

- Up to 16 IP operator consoles can be used on the 3300 ICP system.

- As a rule of thumb, operators will typically handle calls at a rate of 50 to 100CPH. The majority of calls handled by the operator are for incoming trunk traffic.

LAN/WAN Network Configuration

Network Guidelines

To maintain optimum voice quality, voice and data traffic should be separated as much as possible. To separate voice and data traffic, you can

- Run Voice and Data on separate Virtual LANs (VLAN)
- Use a separate subnet for voice traffic
- Use Ethernet switches instead of hubs (voice and data should not use the same shared ethernet hub
- Use Full Duplex Fast Ethernet to the 3300 Controller ports
- Use Full Duplex Fast Ethernet and Ethernet Trunks between switches.

When IP telephones are being placed across routed links, the routers should be configured to prioritize voice traffic by using techniques such as Weighted Fair Queuing (WFQ) with multiple queues configured (for example, high priority for voice and low priority for data). Where the routed connection is across a Wide Area Network (WAN), set the Maximum Transmittable Unit (MTU) appropriately for the speed of the WAN link to minimize delay on slow WAN links.

3300 ICP as a Backup WAN

Many sites have PRI (Primary Rate Interface) access to their telephone system and a separate WAN link for their data. The majority of routers allow for a backup link as well. Instead of renting another BRI or PRI link from the telephone company, you can use the system's existing ISDN access. Using the remote LAN access option on ISDN, configure the card as if it was the interface from the public network; as a result, information can be sent and received from another ISDN-compatible device (such as a router).

Wireless Guidelines

The 3300 ICP provides wireless integration for Symbol wireless devices. To ensure maximum efficiency note the following guidelines:

- Air Access Points (AAP) must be on a single subnet associated with one controller
- AAP connections to one controller must assign the same Electronic Switching System (ESS) number to the devices
- AAP associated with different controllers in the same building must use different ESS numbers.

Features

Features of the 3300 ICP

Feature Name:	Description:
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 to change the COS and COR at any station. 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.
ACD 2000® Extended Agent Groups	Allows you to assign up to 64 agent groups with a maximum of 150 agents in each group.
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.
ACD Make Busy Reason Codes	ACD agents enter a reason code when phones are put into a Make Busy state.
ACD Real Time Event	Makes the ACD Real Time Event stream a purchasable option.
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.
Advanced Analog Networking	Provides calling line identification and travelling class marks across T1/D4 trunks.
Advanced ARS	Allows you to program day and time zones, route plans, and ARS assignment.
Advice of Charge	AOC allows the caller to determine the cost of a toll call.
ANI/DNIS/ISDN Number Delivery	Automatic Number Identification and Dialed Number Identification Service identify numbers that are transmitted on an incoming trunk.
ANSWER PLUS Automatic Attendant	Allows an external caller to dial through to an extension without having to go through an attendant.
ANSWER PLUS Automatic Call Distribution II (ACD 2000)	Consists of four main components: call distribution, agent mobility, management and reporting, and feature configuration and administration.
ANSWER PLUS - Mitel Call Distribution	Permits the use of Recorded Announcement Devices (RADs) and a uniform call distribution to hunt groups.
Attendant Bulletin Board	Bulletin board is shared by all 5550 IP Consoles on the system that have a network connection. Use it to post information for other attendants (for example, speed dial numbers).
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)	Lets you busy-out a specific station by using the attendant console.
Attendant CAS Interface	Centralized Attendant Service interface allows a 3300 ICP to be a

Feature Name:	Description:
	remote node for a CAS site. CAS is an attendant call-handling service provided at a central office switch for calls from both public and private networks.
Attendant Call Answering Priority	Lets you assign priority to calls based on origin when multiple calls are waiting; the call with the highest priority is answered first.
Attendant Call Information Display	Provides the attendant with information about called and calling parties.
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.
Attendant Conference	Lets the attendant set up one or more conference connections between central office trunks and internal stations.
Attendant Consoles (Multiple)	Multiple Attendant Consoles can be supported.
Attendant Console Firmkeys	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	Displays various parameters such as Day/Night Service, Attendant Status, and Alarm Status.
Attendant Directory Number	Lets you dial a number (typically "0") to reach the attendant. Separate directory numbers can be programmed for each attendant console.
Attendant Help	Provides online assistance.
Attendant Hold	Lets you temporarily place a call on hold so you can use other phone features.
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 Language Selection	Enables attendant the to choose the language of operation for the attendant console (English, European French, Canadian French, European Spanish, Latin American Spanish, Dutch, German, Italian, Brazilian Portuguese and European Portuguese).
Attendant Lockout	Prevents the attendant from re-entering a call after releasing it.
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	Lets you use the attendant console to track the cost of outgoing trunk calls.
Attendant New Call Tone	Audio notification of new calls to the attendant console.
Attendant Position Busy-Out	See Attendant Busy-Out (Console).
Attendant Recall	Automatically alerts the attendant when either a trunk call to an idle station or a call on hold at the console has not been answered within a specified time period.
Attendant Ringer Control	Lets you mute the attendant console ringer. When the ringer is muted, the Call Waiting indicator at the top of the display alerts you to incoming calls.
Attendant Scratch Pad	Functions as your personal telephone directory and speed dial list. Use it to save telephone numbers for faster dialing or to store the names and numbers of callers for future reference.
Attendant Serial Call	Automatically returns a call to the attendant console when the call ends.

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Feature Name:	Description:
Attendant Setup and Cancellation of Station Features	Allows the attendant to setup and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder.
Attendant System Login	Requires the attendant to log on to the system to access certain programming functions from the attendant console.
Attendant Tone Signaling	Lets the attendant send tones over the circuit once a call has been established.
Attendant Trunk Group Busy Status	Lets the attendant display and/or print the busy status of the system trunk groups from the attendant console.
Auto-Answer	Automatically answers calls that ring your Prime line. This is typically used in an ACD environment.
Auto-Hold	Automatically places an active call on hold when you press a line key to originate or receive another call.
Automatic Route Selection (ARS)	Simplifies local and long distance dialing by automatically selecting the most convenient and cost-effective route for the call and by inserting and/or deleting the proper routing digits.
Autovon	Allows the system to connect to with Autovon networks (defense switched networks and Canadian switched networks) for incoming and outgoing calls.
BRI (Basic Rate Interface)	A basic ISDN service consisting of two 64Kbps channels and one 16Kbps channel. This feature is supported on 3300 ICP with the 3300 BRI Network Services Unit.
Broadcast Groups	See Groups -Key System and Multicall.
Broker's Call	Lets you temporarily suspend a telephone call while you originate a new one. Once the new call has been established, you can alternate between the two calls.
Busy Dial Through	Lets you dial a feature access code sequence when a busy condition is encountered. See Callback and Camp-on.
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	Lets you program the destination of a matured callback set against a key line or multicall line group.
Call By Name	See Phonebook.
Call Coverage	Provided through a combination of features: Call Rerouting, Call Forward, Do Not Disturb, and Answer Plus™-Mitel Call Distribution.
Call Duration Display	Displays the call duration for incoming and outgoing calls, in one minute increments (starting at 0:00).
Call Forward	Lets you redirect incoming calls to an alternate number.
Call Forward -Cancel All	Lets you cancel all types of Call Forward.
Call Forward -Follow Me-End Chaining	Ensures that calls are not further redirected.
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	Lets you manually redirect an incoming call on your prime or private line to another number.
Call Forward - Override	Lets you bypass or override any Call Forward condition that is set at the station that you are calling.

Feature Name:	Description:
Call Hold	See Hold.
Call Park	Lets the attendant hold a call so that a telephone user can remotely retrieve the call.
Call Pickup	Lets you answer an incoming call that is ringing at another station.
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.
Call Rerouting	Redirects calls to alternate answering points or devices under specified conditions. May be used to redirect calls always (in Day, Night 1, and/or Night 2 mode) or under busy, no answer, or Do Not Disturb conditions.
Call Split	See Conference Split.
Call Swap	See Swap.
Call Transfer	See Transfer.
Call Waiting Swap	Lets you use the switch hook to alternate between two calls when parties are in Call Waiting for your station or when you have a call on Consultation Hold.
Camp-on (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 that a call is waiting. Upon hearing the Call Waiting tone, the busy party can either respond or finish the current call.
Camp-on Tone Security	Prevents you from hearing Camp-on tone. If any party in a call has this option enabled, no Camp-on tone is sent to anyone in the call.
Centralized Attendant Service (CAS) interface	See Attendant CAS Interface
CLASS/CLIP Station Side Software Support.	Enables ONS CLASS/CLIP sets using the CLASS/CLIP protocol to receive caller line identification delivery (CLID) information.
Class of Restriction	Limits a station's access to specified numbers. A station may have three CORs (Day/Night1/Night2 service). The COR may also be changed by using a Verified Account Code.
Class of Service	Defines a station or trunk's feature and timer options. A station or trunk may have three COSs (Day/Night1/Night2 service). The COS may also be changed by using a Verified Account Code.
Clear All Features	Lets you cancel most of the features activated on your extension or another user's extension.
Compression	Allows IP calls to be utilize less bandwidth than an uncompressed call.
Conference	Lets you connect three or more calls into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.
Conference Split	Lets you separate a 3-party conference so that two of the parties can speak privately, while the other is placed on Consultation Hold.
DASS II Voice I	Allows basic calls to be made from the system to a DASS II protocol Central Office, using CEPT Digital Trunks and DASS II signaling.
Date and Time	Set through the System Administration Tool. This data appears on all Station Message Detail Recording (SMDR), traffic

Feature Name:	Description:
	measurements, data dumps, display telephones, and attendant consoles.
Day/Night Service Control	See Night Service.
Dial Tone	You will normally hear continuous dial tone when you lift the handset. You will hear discriminating (also called interrupted), or transfer dial tone under certain conditions.
Dial Tone -Outgoing Calls	The system can provide a pseudo-CO dial tone to prevent possible confusion to station users.
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 fifth 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)	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)	Permits incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system.
Direct Inward System Access (DISA)	Lets external callers access the system by using a special trunk. The system sees the DISA trunk as a station with its own Class of Service and Class of Restriction. Calls that enter the system on DISA trunks have access to a variety of system features. In all cases, the DISA trunk can be assigned account codes to provide a high degree of security or additional options.
Direct Outward Dialing (DOD)	Lets you make external calls without the attendant assistance.
Direct Page	Allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce.
Direct Station Select/Busy Lamp Field (DSS/BLF)	A Busy Lamp Field (BLF) allows the status of a directory number to appear on the line status indicator of a telephone or Programmable Key Module. The monitored device may be on the same system or another system within the same cluster. The key associated with the busy lamp acts as a Direct Station Selection (DSS) key.
Display Contrast Control	Lets you adjust the contrast of the alphanumeric display on your phone.
DNI	Allows the programming of Mitel Networks digital devices
Do Not Disturb	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 a special busy tone.
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.
Emergency Services	Allows an Emergency Services number to be dialed, which sends a Customer Emergency Services ID (CESID) from the system to the Public Safety Answering Point (PSAP). The CESID is used as a key in the Automatic Location Information (ALI) database to retrieve a database record indicating the precise location of the caller.
Feature Keys	Let you activate features without dialing feature access codes.
Flash -Calibrated	Provides an alternate method of generating a Switchhook Flash.

Feature Name:	Description:
Flash -Switchhook	Lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features.
Flash -Trunk	Lets you single- or double-flash a trunk in order to access Centrex™ features.
Flexible Answer Point	Lets station and console users program a night answer point for their incoming trunk calls.
Ground Button	Lets you place a call on Consultation Hold and return to dial tone to invoke station features. The Ground Button provides an alternate method of producing a Switchhook Flash.
Group Page	Lets you page a group of phones over their built-in speakers.
Groups - Key System and Multicall	Let multiple telephones share the same extension number. Incoming calls ring all of the idle stations, and the stations stop ringing when one group member answers the call.
Handset Receiver Volume Control	Lets you adjust the volume of the handset receiver.
Handsfree Operation	Lets you use your telephone without lifting the handset.
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 system through X.409, X.410, and X.25 protocols.
Headset Operation	Lets you use a Headset to make and receive telephone calls.
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 retrieved either at the original answer point or at another extension.
Hold on Hold	Allows both parties of a two-party call to put the call on hold.
Hotel/Motel	Provides a property-management interface and features commonly used by hotels, motels, and hospitals.
Hotline	Automatically dials a designated answer point when you go off-hook. The answer point can be another extension, an attendant, a trunk, or a hunt group.
Hunt Groups	Lets you define a group of extensions under a pilot number; calls to this number ring the first idle extension in the group. You can directly access any phone within a hunt group by dialing it's unique extension number.
Intercept Handling	Lets the system control what happens to a call when it cannot be completed as dialed. Such a call may be routed to a tone or to a directory number; two destinations can be programmed for either condition.
Interconnect Restrictions	Restrict access to certain trunks, stations and equipment (such as data communications equipment). interconnect restrictions are a function of the direction of the call. Every peripheral device is assigned an Interconnect Number that prevents it from connecting with another.
Interconnect Restriction Override	Allows 911 access to telephones in a hotel environment that must be restricted from dialing various internal numbers.
IP Networking	Allows calls to be placed or received over an IP trunk originating from either an IP endpoint or a non-IP endpoint.
Key System Groups	See Groups-Key System and Multicall.

Feature Name:	Description:
Language Change	Lets you change the language of the telephone prompts and softkeys to any one of the following languages: English, French (Canadian), French (European), Italian, German, Spanish (European), Spanish (LA) Dutch, Portuguese (Brazil) or Portuguese (European).
Line Types and Appearances	An administrator can program any of the programmable keys on a phone as line appearance keys for single or shared lines. There are three types of lines: Prime, Non-Prime, and No Where Prime.
Line Appearance Ring Types	Line appearances can be programmed to ring in variety of ways.
Maintenance	The system provides extensive maintenance coverage periodically testing all types of peripheral hardware. Maintenance users may test individual circuits on demand.
Meet Me Answer	Lets a paged party respond to a Group Page without knowing the identity or location of the paging party.
Messaging-Advisory	Displays a short advisory message to display-set users who call your telephone.
Messaging-Callback	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	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.
MNMS	Supports OPS Manager functions.
MSDN/DPNSS	A digital signaling system that provides many features and is used within a private network of PBXs.
MSDN Release Link Trunk	Allows the attendant to make an outgoing call on an incoming trunk. It provides centralized attendant service by allowing attendants on the attendant system to reroute calls without tying up additional trunk resources.
Multicall Groups	See Groups-Key System and Multicall.
Multiple Consoles	See Attendant Consoles (Multiple).
Multiple Message Waiting Indications	Line keys on multiline telephones can be programmed as message waiting indicators that are associated with the mailboxes of other stations.
Multi-Level Auto Attendant	Allows a hierarchical menu to be programmed on the auto attendant providing callers with better self-service access to the person or department they are calling.
Music	Lets you listen to the Music On Hold music source through the speaker on the telephone.
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, is transferred to a busy party, or is in Call Waiting for a station. Customers provides the music source.
Networking	The system supports both analog and digital networking. See Node ID Recognition and Uniform Numbering Plan.
Networked ACD	Supports ACD functions over a Mitel Switched Digital Network (MSDN). Agent groups at different locations (on different systems) may service calls on the network independently of where the call

Feature Name:	Description:
	entered the network.
Networked Group Page	Group Paging can be completed across a network or network cluster, allowing, for example, set on system A to page a specific group on system B.
NI3 Calling Name Delivery	Allows the called party to see the name of a caller on the telephone's display screen if the caller has programmed Calling Name to "Allow" through IMAT. NI3 supports both incoming and outgoing calls for the system T1 card and is supported by the 3300 Universal NSU.
Night Service	Lets you redirect calls to alternate answer points for individual trunks. Answer points can vary, according to the selected mode of operation (Day, Night 1, or Night 2).
Night Service - Automatic	Automatically places the system into Night service if all attendant consoles are unable to receive calls or if all attendant consoles are inactive when the time-out period has expired.
Node ID Recognition	Enables a system in a network to determine whether an incoming call applies to it or to another system in the network.
Non-Busy Station	Lets you program an extension to never return a busy tone. This feature is used for special situations such as emergencies.
Non-DID Extension	Allows the system to support phones that are not directly accessible to DID trunks. Calls to and from these phones are transferred to non-DID extensions by an intercept handling point (such as an attendant or a station).
Off-Hook Voice Announce	Lets you receive a direct page during a handset or headset call. See Direct Page.
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 pulses are sent before the user has finished dialing to allow faster call setup on analog trunks.
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	Prevents users from using Override on your station.
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	Lets you locate and call a system user based by name, extension number, department, and/or location.
PRI (Primary Rate ISDN)	Describes the options supported by the Universal Network Services Unit. These options include Min/Max, Automated Min/Max, NFAS (Non-Facilities Associated Signaling), D-channel Backup and Remote LAN Access.
Printer Support	The system has complete RS-232 printer flexibility. Any printer port may be programmed for any application. The system supports system printers both for its own applications (such as SMDR and maintenance) and as dedicated data communications printers.
Priority Queuing	Handles calls in order of priority. When waiting for calls to be completed internal or external callers are placed in a queue and

Feature Name:	Description:
	assigned an access priority.
Privacy Release	Call privacy between users who share line appearances in key systems groups is automatic. The privacy release feature allows users to release privacy during a call to include another member of the key system group in the call.
Q.SIG	A protocol that allows you to connect a minimum of two systems together to form a virtual private network. Q.SIG supports both incoming and outgoing calls for the 3300 ICP Universal Network Services Unit.
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 are transferred to busy stations or placed on hold.
Recall Button	See Ground Button.
Redial	Automatically dials the last manually dialed number.
Redial -Saved Number	Save a number for future dialing. The number remains saved until a replacement number is saved.
Release	Lets you 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	Lets you program your set to ring and provide a message at a specified time within a 24-hour period.
Remote Wake-up Calls	Wake-up calls can be set or cancelled remotely from a telephone or attendant console using the Hotel/Motel Room Remote Wake-up Call feature access codes.
Ringer Control	Lets you adjust the volume and pitch of the telephone ringer.
Ringing -Discriminating	Lets you distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences).
Ringing -Discriminating (Optional)	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	Lets you answer any ringing line by going off-hook.
SMDR -External	Collects data for outgoing and incoming trunk calls.
SMDR -Internal	Collects data for calls made between stations within the system.
SNMP Agent	Simple Network Management Protocol (SNMP) governs the management and monitoring of network devices and their functions.
Speak@Ease Softkey Support	Provides quick and easy access to the Speak@Ease voice recognition system.
Speaker Volume Control	Lets you adjust the volume of the phone speaker.
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 digit string, the system ceases dialing for the duration of the pause. Dialing resumes when the pause ends
Speed Call -Personal	Lets you store and dial frequently-used numbers using access codes and index numbers.

Feature Name:	Description:
Speed Call -System	Lets you dial stored system numbers.
Speed Dial	See Speed Call.
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 either with a device (device meter unit accumulation) or an account code (account code meter unit accumulation).
Station-To-Station Dialing	Lets you dial any other station directly.
Suite Service	Enables you to group a number of telephone lines through interconnected hotel/motel rooms, or suites, for the purposes of billing and sharing telephone service.
Swap	Lets you temporarily suspend a phone call to originate a new one. Once the new call has been established, you can alternate between the calls.
Switchhook Flash	See Flash-Switchhook.
System Access Authorization	Passwords control administrative access to the system. The installation technician assigns usernames and passwords for access to the different system tools.
System Alarm Indications	See Alarms and Attendant Console Status Display.
System Fail Transfer	Maintains telephone service in the event of system failure (such as during a power outage). When the system goes into SFT mode up to four POTS phones are connected directly to the Central Office. This feature requires a 3300 Universal Analog Services Unit.
T1/D4	Provides support for T1 Channel Associated Signaling. A Dual T1 card is required.
Tag Call	Provides a record of malicious calls in the SMDR record.
TAPI Support	Supports MiTAI and TALK TO TAPI computer telephony interfaces.
Tandem Trunking	The system 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' phones.
Tie Trunk Support	Tie trunks terminate at the attendant console, at station sets, in hunt groups, or on night bells. They may also be arranged as dial-in tie trunks or tandem trunks. Like CO trunks, tie trunks are arranged in groups.
Timed Reminder	See Reminder.
Toll Control	Allows or denies access to specified routes, CO exchanges, and directory numbers.
Tone Demonstration	Lets you hear the tones provided by the system.
Tone Detection	The system 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 system is installed.
Traffic Reporting	Provides traffic reports of system usage to allow better system resource management.
Transfer	Lets you move a call from one phone to another. Before completing

Feature Name:	Description:
	a transfer, you can consult privately with the third party and swap between private conversations with each party.
Transmission Tests	Let you perform milliwatt, balance, and 100 tests on a trunk.
Trunk Access	Lets you directly access a specific trunk. No toll control or ARS checking is done when you use Trunk Access. This feature is used when a maintenance telephone is required
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 that extension.
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 Busy Status	Enables attendants to query the status of trunk groups from the attendant console.
Trunk Group Hunting	Lets you search for trunk groups in either a terminal or circular pattern. In a terminal hunt group, trunks are selected in a predetermined order. In a circular hunt group, trunks are selected in a distributed manner (the first free trunk after the last one used becomes the new first choice).
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	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 trunk access codes.
Trunk Support	The system supports most public network trunk types (both analog and digital).
Uniform Numbering Plan	The system supports the use of a network Uniform Numbering Plan that allows you to use the same digits to reach a station from any location in the network.
Universal Port Orientation	All peripheral interface ports are identical. As a result, the system is flexible and can accommodate various system configurations.
Voice Mail	The system has its own integral voice mail system that supports up to 750 mailboxes.
Voice Mail Interfaces	Most voice processing systems work in conjunction with the system. The system provides the following voice processor interfaces: Voice Mail-Integrated Voice Mail-Digital E&M Interface Voice Mail-ONS Interface
Voice Mail Softkeys	Provides the user with a quick and convenient way to access voice mail. Access to the system is provided through context-sensitive softkeys on an IP telephone.
XNET	Proprietary switched MSDN/DPNSS networking over the PSTN.

Auto Attendant Features

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

Voice Mail Features

Feature	Description
Personal Greetings/Name	Subscriber name and a personal greeting can be recorded by each mailbox user.
Message Prologue	Informs subscribers when they access their mailbox how many new or saved messages they have (if any).
Temporary Greeting	A personal greeting set for a specific number of days (with automatic expiration) can be recorded by each subscriber.
Password Protected Mailboxes	Access to subscriber mailboxes requires a password. Password length system-wide can be from three to six digits. (Default is four digits.) Callers have three chances to enter a valid password before they are disconnected.
Message Envelope	Played prior to beginning of each message, containing priority type, date, and time (including caller identification for internal and external calls). Mailboxes can be individually configured to play the envelope only in response to a key press – i.e., at the request of the subscriber.
Message Length	Unlimited message length with a 5 minute continuation prompt. Minimum message length is two seconds
Saved Messages	Messages may be saved by a subscriber. They will be automatically purged from the system after 15 days (or as reprogrammed) or you can specify that saved messages are never deleted. New messages are never purged automatically. The saved messages are played in last-in first played order
Message Review	Allows immediate replay of a message, including message envelope (timestamp, calling party information).
Message Erase	Allows immediate deletion of a message from the system. The message cannot be subsequently restored; deletion is immediate and permanent.
Message Reply	Allows immediate reply to a message received from another internal mailbox subscriber.
Message Forward	Allows messages to be forwarded to other subscribers and distribution lists with or without a pre-pended comment.
Message Rewind/Hold/Fast Forward	Allows subscribers to rewind, fast forward, or pause messages for several seconds.
Message Keep/Skip	Allows subscribers while listening to a message to advance to the next new message (if any). Each new message played is marked as "saved."
Urgent Messages	The message receives priority placement in the listener's mailbox.
Private Messages	The message cannot be forwarded to another subscriber's mailbox.
Certified Messages	On internal calls, the sender will be notified when the recipient has read the message.
Message Record/Send Actions	Callers will have the ability to pause during recording, review, re-record, and append to a message before sending it. A message can also be cancelled prior to sending.
Message Addressing	Subscribers can address messages to multiple recipients and hear the recipient's name played back to confirm valid entry of mailbox numbers.
Memo	Subscribers will have single-digit access to send a message to their own mailbox, for future reminders and memo-type messaging.

Feature	Description
Message Notification	The subscriber will be notified that they have received a message by the message light on their phone (MWI), and optionally by setting the notification type to one of the following options, which will cause the voice mail system to call: - the mailbox's associated extension number, for analog phone extensions or phones without a message light (prompts called party to log into their mailbox). - an outside number (prompts called party to log into their mailbox). - a message pager (plays an audio message indicating messages are waiting). - a tone-only pager (simply hangs up after a far connection is made). - a digital pager (plays DTMF digits corresponding to a system-wide callback number along with the specific mailbox number). Notification options may be changed by the system administrator. They may also be modified by the mailbox owner if permission is granted by the system administrator. In addition to the notification type, the phone number and schedule are configurable. The schedule determines whether paging occurs: - around the clock, regardless of the business schedule. - only during open business hours. - only during closed business hours. - never (disabled until the schedule is changed to one of the three previous schedule options). Finally, a mailbox may be configured to do non-MWI notification only in response to urgent messages (as opposed to all messages). By default, a busy or no answer condition detected on a notification call will result in two additional retries occurring at 15 minute intervals. All notification results are posted to the system log file.
Outside Message Notification Calls	The administrator will configure a trunk access code for use in all outside notification calls. The trunk access code will control the lines to be used for notification.
Distribution List, Broadcast Message	Allows four system-wide and five (per mailbox) personal distribution lists as well as a broadcast message facility to deliver a message to all mailboxes. Individual subscribers can belong to any number of distribution lists.
New mailbox Tutorial	The system will guide the user through the steps required for initial configuration of mailbox, including specification of a (non-default) passcode and recording of a personal greeting and name.
Mailbox Types	The following mailbox types are available: **Extension** - the auto-attendant will transfer a caller to the mailbox's associated extension. If the called party is busy or does not answer, the caller will be prompted to leave a message in the mailbox. The extension mailbox may be linked to other mailboxes for transfer only (dual mailboxes). This permits the caller to transfer to other mailboxes in the same department. **Message-Only** - the auto-attendant will not attempt a transfer but will immediately prompt the caller to leave a message in the mailbox. **Transfer-Only** - the auto-attendant will transfer a caller to the mailbox's associated extension but will not take a message if the

Feature	Description
	called party is busy or does not answer.
	Information-Only - the auto-attendant will only play the mailbox greeting; no transfer or prompt to leave a message will occur.
	Administrator - for accessing administrative functions such as greetings recording.
Softkey Integration	Users with Mitel Networks telephones can press softkeys instead of dialing codes to select Mitel Express Messenger menu options. For example, to listen to message, a user can press the Play Message softkey instead of dialing the digit 7.
Dual Mailboxes	A transfer-only mailbox can be linked to the same extension as an existing extension-type mailbox. This enables, for example, a single mailbox for a sales department and the sales manager.
Mailbox Administration via OPS Manager	Mailbox administration (adds, moves, changes) can be performed using Mitel Networks OPS Manager, a standalone application that works seamlessly with the 3300 ICP embedded system management.

Product Availability by Region

Asia Pacific

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

Asia Pacific		
	Australia	**New Zealand**
3300 ICP Components	Y	Y
3300 Voice Mail	Y	Y
3300 Wireless	Y	Y
Digital Services Unit	N	Y
Peripheral Unit	N	Y
Applications		
OPS Manager*	Y	Y
5700 Voice First Applications	Y	Y
5810 PDA Application	Y	Y
6110 Contact Center Management	Y	Y
6115 Interactive Contact Center	Y	Y
6120 Contact Center Scheduling	Y	Y
6150 Multimedia Contact Center	Y	Y
6160 Intelligent Queue	Y	Y
6200 Cost Management	Y	Y
6300 Call Recording	Y	Y
6500/6510 Unified Communications	Y	Y
6600 Your Assistant	Y	Y
Manual Maker*	Y	Y
* Note: This application is supported but the interface is in English only.		
Peripherals		
5001 IP Phone	Y	Y
5005 IP Phone	Y	Y
5010 IP Phone	Y	Y
5020 IP Phone	Y	Y
5140 IP Appliance	Y	Y
5303 Conference Phone	N	N
5305 IP Office Conference Unit	Y	Y
5310 IP Board Room Conference Unit	Y	Y
5201 IP Phone	Y	Y
5205 IP Phone	Y	Y
5215 IP Phone	Y	Y

Asia Pacific		
	Australia	**New Zealand**
5220 IP Phone	Y	Y
5230 IP Appliance	Y	Y
5240 IP Appliance	Y	Y
Symbol MiNET Wireless Phone**	Y	Y
SUPERSET 4001	N	Y
SUPERSET 4015	N	Y
SUPERSET 4025	N	Y
SUPERSET 4150	N	Y
**Note: Symbol devices support an English interface only.		
Consoles		
SUPERCONSOLE 1000	N	Y
5550 IP Console	Y	Y
Accessories		
5410 PKM	Y	Y
5415 PKM	Y	Y
IrDA Module		
5412 PKM	Y	Y
5448 PKM	Y	Y
5424 IrDA Module	Y	Y
5485 IP Paging Unit	Y	Y
3300 Power Dongle	Y	Y
PowerDsine 24PT In-line Power Unit	Y	Y

EMEA Region

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

EMEA Region									
	UK	Spain	Portugal	Netherlands	Italy	Germany	France	UAE	South Africa
3300 ICP Components	Y	Y	Y	Y	Y	Y	Y	Y	Y (no Universal ASU)
3300 Voice Mail	Y	Y	Y	Y	Y	Y	Y	Y	Y
3300 Wireless	Y	Y	Y	Y	Y	Y	Y	Y	Y
Digital Services Unit	Y	N	N	Y	Y	Y	N	Y	Y
Peripheral Unit	Y	N	N	Y	Y	Y	N	Y	Y
Applications									
OPS Manager*	Y	Y	Y	Y	Y	Y	Y	Y	Y
5700 Voice First Applications	Y	Y	Y	Y	Y	Y	Y	N	N
5810 PDA Application	N	N	N	N	N	N	N	N	N
6110 Contact Center Management	Y	N	N	Y	Y	N	N	N	N
6115 Interactive Contact Center	Y	N	N	N	N	N	N	N	N
6120 Contact Center Scheduling	Y	N	N	N	N	N	N	N	N
6150 Multimedia Contact Center	Y	N	N	N	N	N	N	N	N
6160 Intelligent Queue	Y	N	N	N	N	N	N	N	N
6200 Cost Management	N	N	N	N	N	N	N	N	N
6300 Call Recording	N	N	N	N	N	N	N	N	N
6500/6510 Unified Communications	Y	N	N	N	N	N	N	N	Y
6600 Your Assistant	Y	Y	Y	Y	Y	Y	Y	Y	Y
9100 Call Center Commander	Y	N	N	N	N	N	N	N	N
Manual Maker*	Y	N	N	N	N	N	N	N	N
* Note: This application is supported but the interface is in English only.									
Peripherals									

EMEA Region									
	UK	Spain	Portugal	Netherlands	Italy	Germany	France	UAE	South Africa
5001 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5005 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5010 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5020 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5140 IP Appliance	Y	Y	Y	Y	Y	Y	Y	Y	Y
5303 Conference Phone	Y	N	N	N	N	N	N	N	N
5305 IP Office Conference Unit	N	N	N	N	N	N	N	N	N
5310 IP Board Room Conference Unit	Y	Y	Y	Y	Y	Y	Y	Y	Y
5201 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5205 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5215 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5220 IP Phone	Y	Y	Y	Y	Y	Y	Y	Y	Y
5230 IP Appliance	Y	Y	Y	Y	Y	Y	Y	Y	Y
5240 IP Appliance	Y	Y	Y	Y	Y	Y	Y	Y	Y
Symbol MiNET Wireless Phone**	Y	Y	Y	Y	Y	Y	Y	Y	Y
SUPERSET 4001	Y	N	N	Y	Y	Y	N	Y	Y
SUPERSET 4015	Y	N	N	Y	Y	Y	N	Y	Y
SUPERSET 4025	Y	N	N	Y	Y	Y	N	Y	Y
SUPERSET 4150	Y	N	N	Y	Y	Y	N	Y	Y
**Note: Symbol devices support an English interface only.									
Consoles									
SUPERCONSOLE 1000	Y	N	N	Y	Y	Y	N	Y	Y
5550 IP Console	Y	Y	Y	Y	Y	Y	Y	Y***	Y
*** Note: This application is not available in Arabic									
Accessories									
5410 PKM	Y	Y	Y	Y	Y	Y	Y	Y	Y
5415 PKM	Y	Y	Y	Y	Y	Y	Y	Y	Y
5423 IrDA Module	N	N	N	N	N	N	N	N	N
5412 PKM	Y	Y	Y	Y	Y	Y	Y	Y	Y
5448 PKM	Y	Y	Y	Y	Y	Y	Y	Y	Y
5424 IrDA	N	N	N	N	N	N	N	N	N

EMEA Region									
	UK	**Spain**	**Portugal**	**Netherlands**	**Italy**	**Germany**	**France**	**UAE**	**South Africa**
Module									
5485 IP Paging Unit	N	N	N	N	N	N	N	N	N
3300 Power Dongle	Y	Y	Y	Y	Y	Y	Y	Y	Y

Latin America Region

This table is a list of products available in Latin America. It does not necessarily denote that the product has completed regulatory approvals.

Latin America				
	Argentina	**Brazil**	**Chile**	**Mexico**
3300 ICP Components	Y	Y	Y	Y
3300 Voice Mail	Y	Y	Y	Y
3300 Wireless	Y	Y	Y	Y
Digital Services Unit*	N	N	N	N
Peripheral Unit*	Y	Y	Y	Y
Note: These components are supported from a migration perspective.				
Applications				
OPS Manager*	Y	Y	Y	Y
5700 Voice First Applications	Y	Y	Y	Y
5810 PDA Application	N	N	N	N
6110 Contact Center Management	Y	Y	Y	Y
6115 Interactive Contact Center	Y	Y	Y	Y
6120 Contact Center Scheduling	Y	Y	Y	Y
6150 Multimedia Contact Center	Y	Y	Y	Y
6160 Intelligent Queue	Y	Y	Y	Y
6200 Cost Management	Y	Y	Y	Y
6300 Call Recording	Y	Y	Y	Y
6500 Unified Communications (attendant only)	Y	Y	Y	Y
6600 Your Assistant	Y	Y	Y	Y
Manual Maker*	Y	Y	Y	Y
* Note: This application is supported but the interface is in English only.				
Peripherals				
5001 IP Phone	Y	Y	Y	Y
5005 IP Phone	Y	Y	Y	Y
5010 IP Phone	Y	Y	Y	Y

Latin America				
	Argentina	**Brazil**	**Chile**	**Mexico**
5020 IP Phone	Y	Y	Y	Y
5140 IP Appliance	Y	Y	Y	Y
5303 Conference Phone	Y	Y	Y	Y
5305 IP Office Conference Unit	Y	Y	Y	Y
5310 IP Board Room Conference Unit	Y	Y	Y	Y
5201 IP Phone	Y	Y	Y	Y
5205 IP Phone	Y	Y	Y	Y
5215 IP Phone	Y	Y	Y	Y
5220 IP Phone	Y	Y	Y	Y
5230 IP Appliance	??			
5240 IP Appliance	Y	Y	Y	Y
Symbol MiNET Wireless Phone**	Y	Y	Y	Y
SUPERSET 4001	Y	Y	Y	Y
SUPERSET 4015	Y	Y	Y	Y
SUPERSET 4025	Y	Y	Y	Y
SUPERSET 4150	Y	Y	Y	Y
**Note: Symbol devices support an English interface only.				
Consoles				
SUPERCONSOLE 1000	Y	Y	Y	Y
5550 IP Console	Y	Y	Y	Y
Accessories				
5410 PKM	Y	Y	Y	Y
5415 PKM	Y	Y	Y	Y
IrDA Module	Y	Y	Y	Y
5412 PKM	Y	Y	Y	Y
5448 PKM	Y	Y	Y	Y
5424 IrDA Module	Y	Y	Y	Y
5485 IP Paging Unit	Y	Y	Y	Y
3300 Power Dongle	Y	Y	Y	Y

Solutions

Overview

In this section, examples of customer requirements illustrate 3300 ICP and Mitel Networks applications solutions for examples for the following three scenarios:

- Stand-alone site
- Multiple system site
- Installed base

Stand-alone Site

The Customer

Really Good Autoparts (RGA) manufactures auto parts for a Big Three car manufacturer in Detroit. Since being bought out a year and a half ago, RGA has become a dynamic company that is known for its high quality and excellent turnaround time. RGA runs an integrated facility with manufacturing, marketing, and sales located in the same building.

The Problem

As with any large manufacturing operation, efficiency and cost containment are essential to RGA's continuing success. As a result, RGA management is concerned by several communications problems the company is experiencing. First, sales people are having difficulty prioritizing the volume of voice messages that they are receive resulting in customer complaints about response time. Second, troubleshooting the production lines is becoming more expensive as volume and complexity increase. Third, customer service is having trouble keeping up with orders and support calls to satisfy the demands of this RGA's growing business.

The Solution
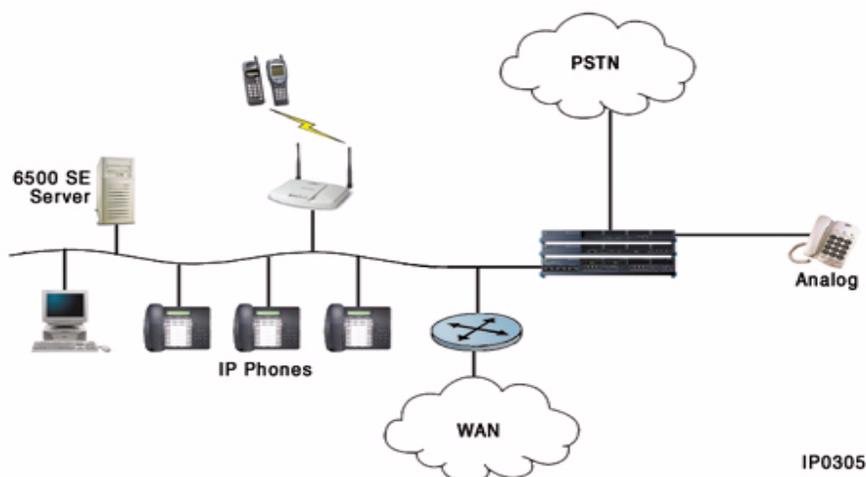
Management teams with a Mitel Networks VAR to evaluate and implement an integrated communications solution that will meet current and future needs.. The foundation of the solution is the Mitel Networks 3300 Integrated Communications Platform (ICP). Employing leading-edge Voice-over-IP (VoIP) technology and supporting both traditional and innovative call features and applications, the 3300 ICP provides voice communications systems that are equal or superior to today's best PBXs. Adaptable and secure the 3300 ICP exemplifies the reliability upon which Mitel Networks has built its reputation for almost 30 years.

As part of the its 3300 ICP solution, RGA selects the Mitel Networks 6500 Unified Communications to enable the sales team to ensure customer satisfaction by efficiently managing voice, email, and fax messages. With a single message store, the 6500 Unified Messaging combines voice mail, email, and faxes in one inbox, which users can navigate with natural speech commands. This ability provides users with the flexibility to manage messages by sender, date, or type; forward or reply to the messages with voice; or to simply return a call without having to look up the number. To further enhance productivity, users can check the calendar, make appointments and meeting requests, and create tasks through the voice user interface. Managing messages in this way while on the road or in the office affords RGA'a sales team the advantage of conducting business anywhere, anytime.

Next, to facilitate and cut costs of troubleshooting assembly lines and desktops, RGA decides to provide in-building mobility to its technicians. The 3300 ICP provides full communications mobility by supporting Symbol Spectrum24 and NetVision, to make and receive calls from anywhere in their facility to consult with colleagues about a problem as they are working to resolve it. This mobility ensures efficient problem resolution and helps RGA spend less on technician hours. Moreover, because the system is IP based, technicians can easily check the trouble ticket database to ensure that the next ticket they solve is a high priority; thus, their time is spent primarily on problems with the highest business impact.

RGA benefits from the integrated ACD functionality of the 3300 ICP, combined with the Mitel Networks 6100 Contact Center Solutions (CCS) applications, which provides management with the tools to efficiently manage their contact center and solve their customer service difficulties. The Mitel Networks 6110 Contact Center Management (CCM) application keeps managers abreast of issues affecting service in real time. In conjunction with the 6110 CCM, the Mitel Networks 6115 Interactive Contact Center application is the perfect solution to manage RGA's multiple queues and erratic call volumes. Through a Web browser, supervisors can remove ACD queues from service in times of low call volume, and return them to service when inbound call volume increases. Using the Mitel Networks 6120 Contact Center Scheduling solution, management can create customized schedules for agents to ensure that the right amount of staff maintains their required service levels.

Because of its modular design, the 3300 ICP can keep up with RGA as their business grows by clustering 3300 ICP controllers to support thousands of local and/or remote users.
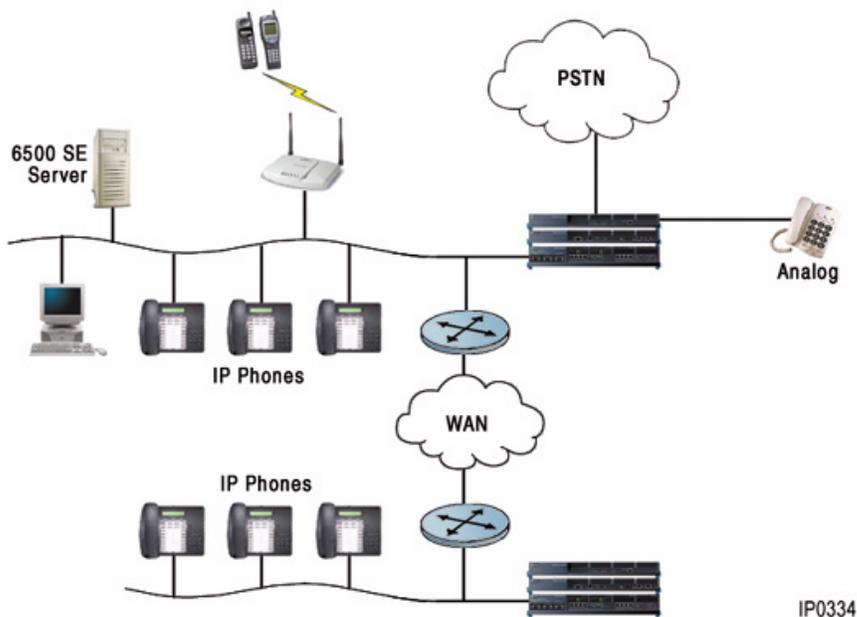
Multiple Site

The Customer

Excellent Advertising (EA) is a Park Avenue advertising firm with a reputation for innovative, advertising campaigns. In addition to its head office in New York, the company has a large branch office in San Francisco that takes care of its West Coast business. The nature of the advertising business requires that large files be transferred between EA's East and West Coast offices. A robust corporate WAN has been put in place to support this need.

The Problem

Excellent Advertising (EA) is concerned that its business and costs are out of control: the company's long-distance bills are spiraling upwards, and the management of two independent PBXs is becoming increasingly difficult; employees are complaining about wasting time dialing network access codes to call their colleagues on the other side of the country; and on the road, the sales team can't keep up with messages left by clients. As a result, the managing partners have decided that their legacy telecommunications equipment is restricting their success. They decide it's time for a change.

The Solution

IP0308
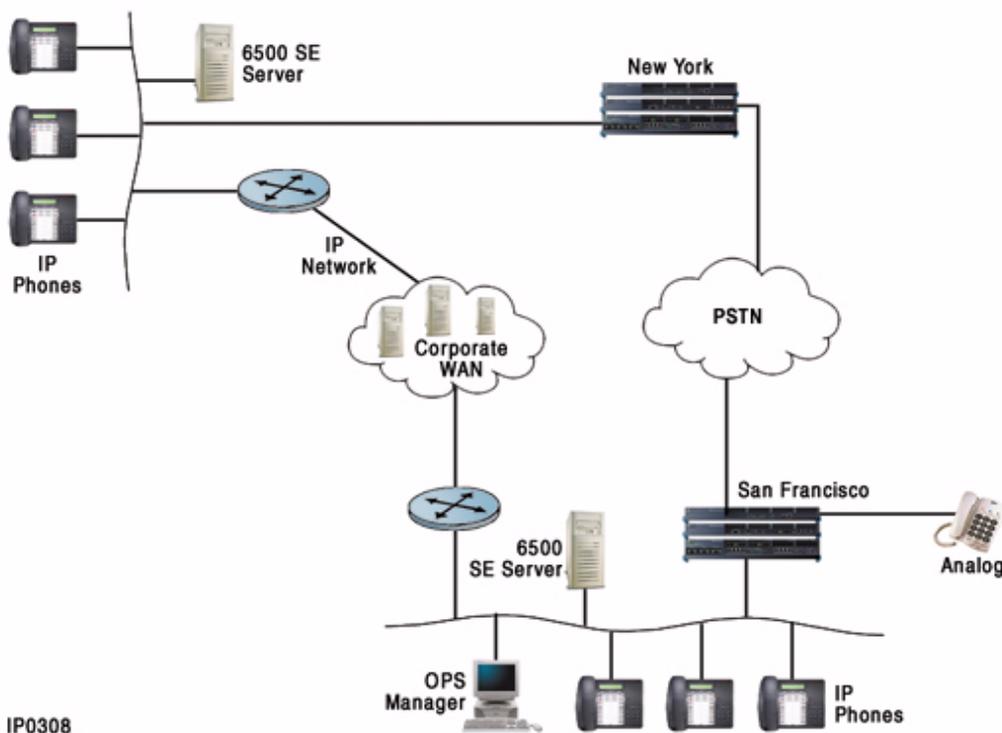
EA's communications service provider recommends a solution built around the Mitel Networks 3300 ICP. The 3300 ICP supports flexible networking and natural- speech recognition technology, both of which deliver tangible business advantages.

EA's solution begins with installing two 3300 ICPs, one in the New York headquarters and one in the San Francisco branch office. The two 3300 ICPs are connected through the integrated IP Networking feature of the 3300 ICP, thus leveraging the corporate WAN. By sending voice traffic over the corporate WAN, EA realizes significant long distance savings

Since the full suite of MSDN features is available over IP networking, EA is able to implement a four-digit dialing plan that allows simple and efficient cross-country calling. In addition, the company is able to take advantage of other MSDN features like calling line ID, callback, and call pickup.

Since the 3300 ICP supports all of Mitel Networks' legacy features, ARS is used to route network calls over the PSTN as a backup in the event of congestion on the corporate WAN.

OPS Manager is installed in EA's New York headquarters to manage the network. OPS Manager helps reduce operating costs by simplifying day-to-day system administration and network maintenance tasks such as moves/adds/changes, directory management, alarms management, database backups, remote maintenance, and more. With OPS Manager, EA management has centralized control over all elements of the company's communications system, anytime, anywhere.

Mitel Networks 6500 Unified Messaging lets users selectively navigate their unified inbox using natural voice commands. Sales staff calling in to the unified messaging system can reply to, forward, and return calls and messages from any location. This flexibility allows EA sales staff to immediately reply to an urgent message from a valued client instead of having to answer all messages in the order that they are recorded. The ability to prioritize calls with 6500 Unified Messaging results in superior customer service and client satisfaction.

Installed Base Migration

The Customer

Really Good Real Estate (RGRE) is a Toronto- based company that provides commercial real estate management and leasing. The business has seen steady growth in the past five years, and now management has decided to expand to Europe.

The Problem

Initially, RGRE will open an office in London. Management is concerned that they may have trouble managing the costs of their telecommunications infrastructure. Since most of their experienced staff will remain in Canada, they expect a high volume of overseas calls. RGRE owns a Mitel Networks SX-2000 and wants to ensure a continuing return on their investment in this system as the company expands.

The Solution

IP0310
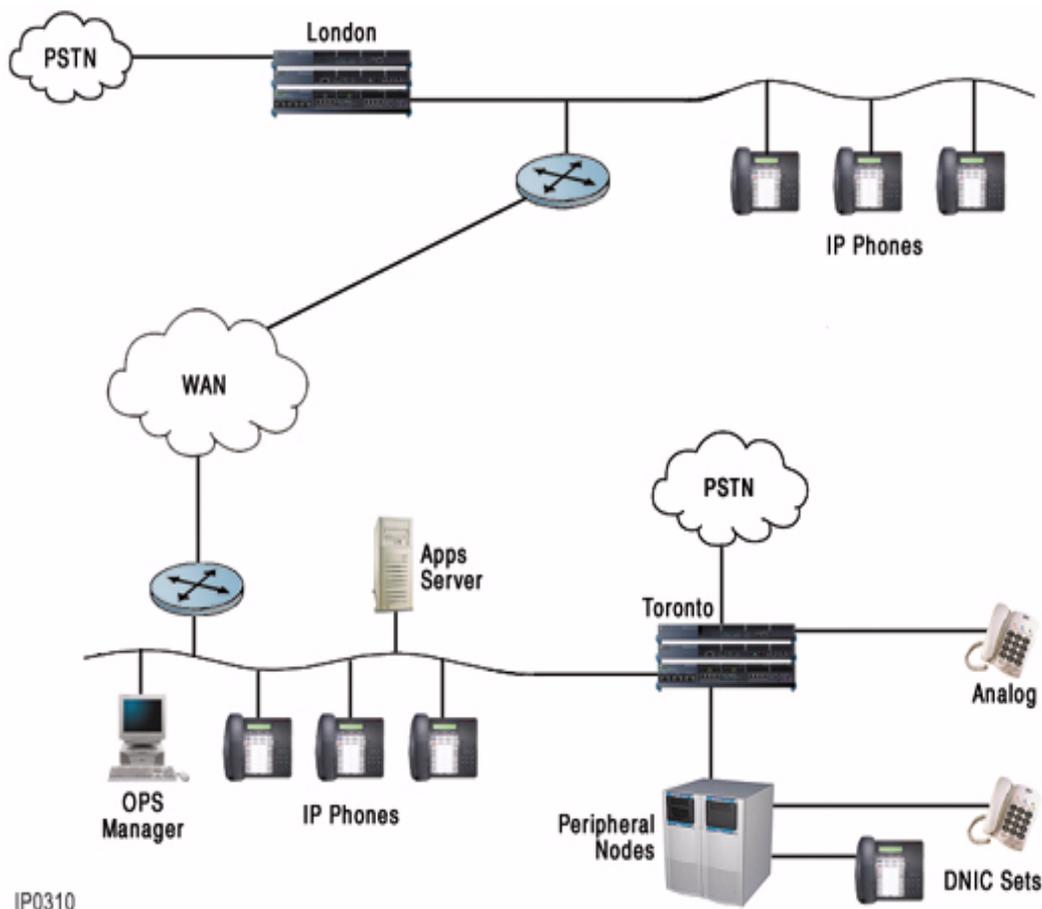
Central to RGRE's solution is the Mitel Networks 3300 ICP. The new switch in London will be a 3300 ICP with integrated voice mail. This system will provide high-quality voice communications and advanced, integrated applications that scale as needed. Since the 3300 ICP is a VoIP system, RGRE will save money wiring their new office in London—only one set of wiring needs to be installed for both the voice and data network.

In order to save on costly PSTN long-distance charges, RGRE's service provider decides to route all long-distance traffic over the corporate WAN, something which is easily accommodated by the 3300 ICP's integrated IP Networking feature. In the Toronto office, RGRE installs a 3300 ICP, which will be connected to the 3300 ICP in London via IP Networking. In addition to realizing significant cost savings with this setup, RGRE will be able to take advantage of MSDN features across the network.

Due to the flexibility of the 3300 ICP, SX-2000 peripherals are supported on the new platform. The peripherals on the SX-2000 in Toronto will migrate and be connected to the 3300 ICP. These peripherals will continue to support time division multiplexing (TDM) phones, providing investment protection for the existing equipment. As RGRE's business grows, new IP phones can be installed on the 3300 ICP.

OPS Manager will be installed in Toronto to centrally manage the network. OPS Manager helps reduce operating costs by simplifying day-to-day system administration and network maintenance tasks such as station set moves/adds/changes, directory management, alarms management, database backups, remote maintenance and more. RGRE managers will have centralized control over all system elements, anytime, anywhere.

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