MITEL

MITEL NETWORKS

3300

Integrated Communications Platform

General Information Guide

Release 3.1

Introduction

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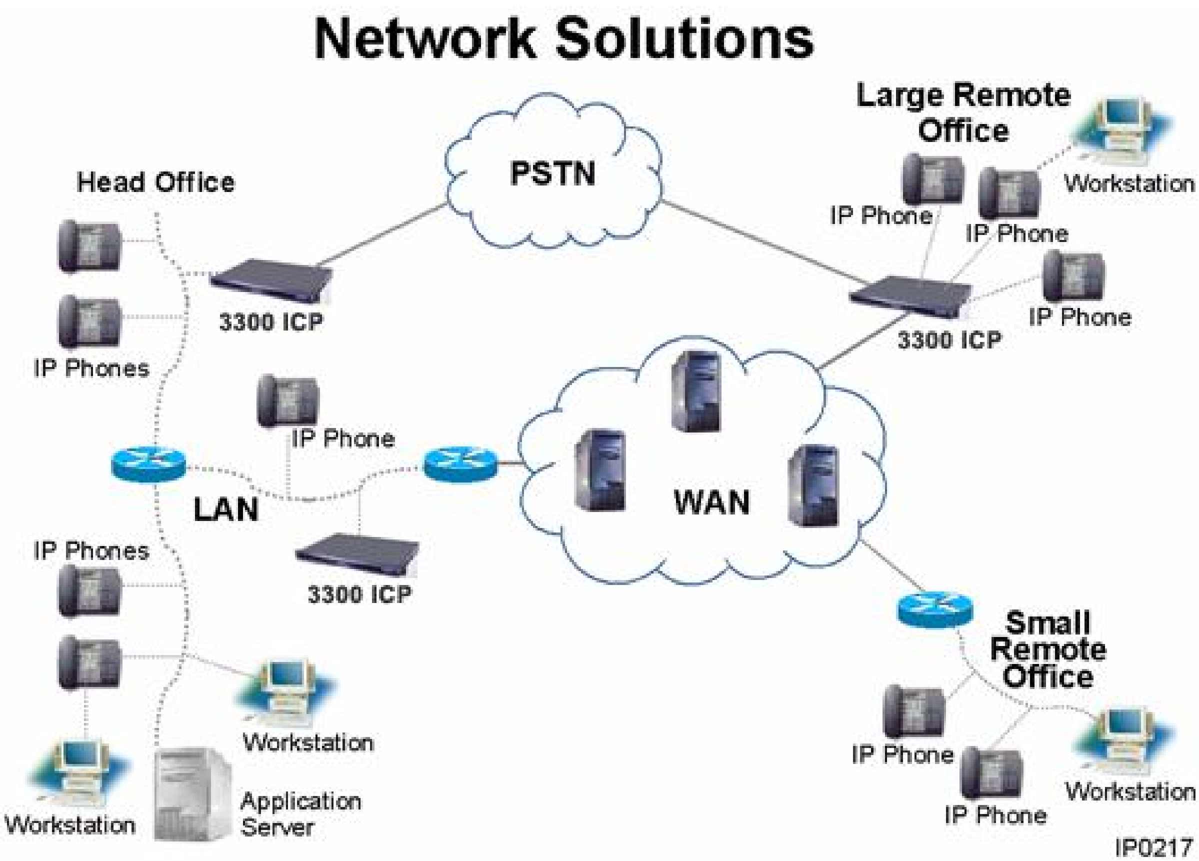
Overview

Product Overview

The Mitel Networks™ 3300 Integrated Communications Platform (ICP) is a Voice over IP (VoIP) solution for medium to large enterprises and for corporate enterprises with branch offices. The Mitel Networks 3300 ICP provides a complete solution by delivering sophisticated call management, desktop peripherals, and applications. All of these components are easily managed by a web-based application.

A single 3300 ICP supports up to 700 IP telephones per system. With the ability to cluster multiple systems together, the system supports thousands of users (whether they are co-located or geographically dispersed). The clustered systems can all be managed as a single network from any location.



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. This approach uses the IP network for connecting IP telephony devices and provides a supplementary TDM (Time Division Multiplexing) bus for switching calls between traditional telephone devices. Using this approach all traffic is switched optimally as traffic is connected in it's native format.

The 3300 ICP controller provides call setup, tear down, and signaling between Ethernet IP connected telephones. For traditional telephony, such as POTS and PSTN trunks, non-IP circuit communications switching is handled via the conventional TDM circuit-switched bus.

This ability to use two different switching techniques simultaneously means that

- Gateway functionality is only required between the IP and non-IP networks
- All traffic is switched with minimum conversion between packet and traditional telephony to provide optimum voice quality in all call scenarios.

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

Mitel Networks 3300 Controller

The Mitel Networks 3300 Controller provides the voice, signaling, central processing, and communications resources for the system. It also houses an optional integrated voice mail system.

The 3300 Controller contains a hard drive, two RS-232 ports (DB-9 connectors), a remote alarm port, 64 or 128 channels of echo cancellation, a Stratum 3 clock, and a power-fail-protected real-time clock. For specific details on the 3300 Controller hardware, refer to the Hardware User Guide.

The 3300 Controller connects to Network Services Units (NSUs) using multi-mode fiber terminated on an ST connector. The 3300 Controller connects to Analog Services Units (ASUs) by using copper cables terminated on an RJ-45 connector. The 3300 Controller connects to the customer's LAN and IP devices by using the four 10/100BaseT Ethernet switch ports located on the front panel via the customers LAN.

In addition existing SX-2000 Digital Services Units (DSUs) and Peripheral cabinets can be connected over a fiber cable.

Two sizes of 3300 Controller are available. These units provide direct connectivity for either

- 2 NSUs direct connection (4 NSUs if chaining is used) 4 ASUs, 16 Attendant Consoles, 250 IP telephones and 64 echo cancellers
- 4 NSUs direct connection (8 NSUs if chaining is used), 4 ASUs, 16 Attendant Consoles, 700 IP telephones and 128 echo cancellers.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Network Services Units

Network Services Units

The Network Services Units (NSUs) provide 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 (NSU)
- Mitel Networks 3300 R2 Network Services Unit (NSU)
- Mitel Networks 3300 BRI Network Services Unit (NSU).

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 and 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 by using 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.


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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 by using a fiber cable.

Additional R2 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. It is available in North American and European variants. The North American variant supports user-side interfaces. 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 to a 3300 Controller directly but instead connects to the Mitel Networks 3300 Universal NSU by using a copper cable. For detailed information on the hardware specifications, refer to the Hardware User Guide.

Analog Services Units

Analog Services Units

The Analog Services Units (ASUs) 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 (ASU)
- Mitel Networks 3300 Analog Services Unit (ASU).

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

Mitel Networks 3300 Universal Analog Services Unit

The Mitel Networks 3300 Universal ASU is the combination unit that houses 16 ONS CLASS (Custom Local Area Signaling System) and four Loop Start (LS) trunk CLASS 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 of the unit. The unit connects to a 3300 Controller by using a Category 5 Universal Twisted Pair (UTP) cable that terminates on an 8-pin modular jack (RJ-45).

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

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 cable that terminates on an 8-pin modular jack (RJ-45).

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

Migration

Migration

The 3300 ICP offers the ability to migrate from an existing SX-2000 MICRO LIGHT, SX-2000 LIGHT, or 3200 ICP. The database 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 main cabinet, you can physically connect it to the 3300 ICP by using multi-mode fiber. As a result, you can use the existing peripheral and digital trunk cards within the main cabinet. Any external cabinets can also be connected by using FIMs.

Mitel Networks SX-2000 LIGHT

The SX-2000 LIGHT Digital Services Unit (DSU) cabinet provides digital trunk capability, and the SX-2000 LIGHT peripheral cabinet provides connectivity for analog trunks, analog telephones, and Mitel Networks DNI devices. Both cabinet types can be connected to the 3300 Controller by using 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, and 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 connected to the FIM ports on the 3300 Controller.

In addition to the DNIC telephones supported by the peripheral cabinet, the 3300 ICP supports the following legacy single port IP telephones:

- SUPERSET 4015IP
- SUPERSET 4025IP.

Maintenance

Alarms

An alarm is an event that takes place when an anomaly is detected and corrective action is required. All attendants who use the Mitel Networks consoles are provided with alarm status information when an alarm is raised. 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 needs 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 repair person, but it is not urgent. Examples of a minor alarm include the loss of a single line or trunk circuit.

An alarm condition is cleared when the fault is corrected.

Circuit Indicators

The system has Light Emitting Diodes (LEDs) on the front of each component 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:

- System Administration Tool provides a web-based interface that trained technicians use to program the system. It requires Microsoft Internet Explorer 6.0 or later.
- 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 later.
- Desktop Tool provides a web-based interface that enables IP telephone users to program their telephone feature keys. It requires Microsoft Internet Explorer 5.5 or later.
- 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 systems, and 3200 ICP systems to be migrated forward to the 3300 ICP by using a database conversion and restore utility. The 3300 Configuration Tool requires Microsoft Windows NT 4.0 or Microsoft Windows 2000 Professional operating system.
- 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’s important to assign appropriate telephone privileges to users and devices. In addition, telephones in public places (such as a lobby telephone) 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 being connected together
- 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 giving detailed information including the originating extension number, time, duration, and number dialed. SMDR record access should be restricted as with any other function.

Authorized Maintenance 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 and these passwords should be changed periodically.

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

Specifications

Environment

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)
Conversion factors: 1 watt is equal to 3.412 BTUs per hour, 1 ton of refrigeration is equal to 12,000 BTUs per hour or 3.516 Kilowatts, and 3/4 Kilowatt-hour is equal to 1 ton of refrigeration.	

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 (4.53kg)

Tone Plans

The system supports tone plans for the following countries:

- North America
- United Kingdom
- Latin America (Argentina, Chile, Mexico).

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) is supported 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 (ACD)

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

The system also supports ACD functions over the MSDN/DPNSS digital network 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 by using the Mitel Networks 6100 Contact Center Solutions product line.

Voice Mail

The 3300 ICP includes a complete voice mail system designed to improve communication between your company, clients, customers, and employees. Programming is completed by using the System Administration Tool. Up to 20 ports are available for voice mail calls with support for a maximum of 750 mailboxes and 100 hours of storage time. Support is also available for attendant functionality such as automated attendant. The voice mail system supports English, French, Spanish, and Dutch languages.

Features provided by the voice mail system include:

- 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
- Subscriber mailboxes that are password-protected
- 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.

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
- Integrate the network telephone directory with a network directory service database
- Schedule automatic upgrades, database saves, and database restores
- 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 4.05 browser or a Microsoft Internet Explorer browser from any Windows NT, Windows 95/98, or Windows 2000 workstation on the network.

Mitel Networks 6200 Cost Management Solutions

The Mitel Networks 6200 Cost Management Solutions provide end users with useful facts about information travelling on their voice and data networks. Using this information, companies can increase efficiency and reduce costs.

The 6200 Cost Management Solutions are an application portfolio for monitoring and managing communication resources. Mitel Networks has two Cost Management Solutions available.

- Mitel Networks 6210 Cost Management General Business Call Accounting
- Mitel Networks 6220 Cost Management Hospital Call Accounting - Hospitality, Education, Public Health.

Optional Modules include:

- LDAP: Lightweight Directory Access Protocol (LDAP) enables synchronization of directories from various systems.
- Enhanced Data Collection Utility: Using this utility, data can be collected directly from the 3300 ICP over the LAN.
- Optional Hardware - MDR-2000e Ethernet ISD: The MDR-2000e collects, stores, and downloads information from remote or local telephone systems. It uses TCP/IP protocol to poll across your network. This microprocessor-based system can simultaneously handle call detail recording translation, alarm monitoring, and traffic data. The MDR-2000e offers increased polling efficiency, decreased equipment costs, and cost savings on remote polling long distance charges.

Note: Optional hardware may be required depending on configuration.

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 contact centers with the tools they need to record calls for liability purposes and/or evaluate the quality of agents' work, measure performance and improve the response of the agents.

Features of this solution include:

- Call Recording
- Bomb threat/Abusive call selective save
- Last Call Repeat Facility
- Instant Recall Terminal
- Quality Assurance (browser-based application overlays recording platform).

For detailed information refer to Mitel Networks 6300 Call Recording and Quality Monitoring Solution documentation.

Mitel Networks 6500 Speech-Enabled Applications

Mitel Networks 6500 Speech-Enabled Applications

Mitel Networks 6500 Speech-Enabled (SE) Applications deliver powerful and unique solutions. The system responds to conversational voice commands and provides a single point of access for a wide range of information.

A Mitel Networks 6500 Speech Enabled Applications server can support from 2 to 16 ports of simultaneous speech, corporate directories with up to 10,000 names, and personal directories of up to 500 names. The server connects to the system through the LAN by using the IP protocol.

6500 SE Applications run on Microsoft Windows NT Server 4.0

There are two main applications:

- Mitel Networks 6500 Speech-Enabled Attendant
- Mitel Networks 6500 Speech-Enabled Unified Messaging.

There are also four additional purchasable options available:

- Enterprise Voice Portal
- Mobility
- Calendar and Task Management
- Fax Integration.

Mitel Networks 6500 Speech-Enabled Attendant

The Mitel Networks 6500 Speech-Enabled 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.

Using this application

- Users can place a call to any number in the company directory by stating a name, extension, or department
- Users can call into the system from their home telephone or cellular telephone and place calls to external numbers that are programmed in the company directory provided they have been assigned the required system privileges

- Registered users can program their own list of frequently called numbers and then place calls to those numbers by using speech.

Mitel Networks 6500 Speech-Enabled Unified Messaging

Mitel Networks 6500 Speech-Enabled Unified Messaging is fully integrated with Microsoft Outlook and the Mitel Networks 6500 Speech Enabled Attendant application. It provides access to all messages from a single Outlook Inbox that users can filter and navigate via speech recognition.

Using this application, users can

- Ask the system to play messages based on caller, date, type, and priority without having to view messages sequentially
- Dial calls by simply saying the contact’s name (through access to Outlook's contact list).

Mitel Networks 6500 Speech-Enabled Applications - Options

These applications are available as additional purchasable options on the 6500 SE Attendant or 6500 SE Unified Messaging. For complete details, please refer to the Mitel Networks 6500 Speech-Enabled Applications documentation.

Enterprise Voice Portal

Enterprise Voice Portal uses VoiceXML (Voice Extensible Markup Language) technology that allows interaction with the Internet 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 surf a web-site by using voice commands and link to other related sites.

Mobility

The Mobility feature provides users with the ability to redirect the 6500 SE 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 users default are directed to the "reach me at" number. However, if a caller specifically requests the users cellular phone, pager, facsimile, or home phone, then the call is directed to the requested number.

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, create, modify, or delete appointments, meetings or tasks.

Fax Integration

Fax Integration provides users with the ability to sort their message mail by type and can forward their faxes to another fax machine. The faxes are stored on the fax server and the Exchange 2000 Server.

Mitel Networks 6100 Contact Center Solutions

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 incorporates

- 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 Contact Center Quality Monitoring
- Mitel Networks 6180 Contact Media Costing.

Mitel Networks 6110 Contact Center Management

Mitel Networks 6110 Contact Center Management is an application that manages ACD information. It uses the familiar Microsoft Excel and Microsoft Internet Explorer interface. Contact center managers can log on to any PC to run reports, monitor real time activities, forecast the number of agents required, and perform numerous management functions over the network.

The incorporated applications include:

- Reporter
- Report Inbox
- SuperAdvisor
- AgentAdvisor
- SuperAuditor
- Wallboarder
- Forecasting
- Multisite Reporting, Monitoring and Forecasting
- Chatline
- Inspector.

Mitel Networks 6115 Interactive Contact Center

Mitel Networks 6115 Interactive Contact Center enables supervisors to alter the way calls are handled in the contact center. The web-based interface enables supervisors to

- Put queues (paths) in and out of operation (DND)
- Schedule queues to open or close based on time of business day
- Put Agents in and out of DND
- Put Agents in and out of Make Busy
- Log in and log out Agents
- Move Agents to different answer points
- Agents can make busy with reason codes.

Mitel Networks 6120 Contact Center Scheduling

Mitel Networks 6120 Contact Center Scheduling works in conjunction with the Mitel Networks 6110 Contact Center Management application enabling the optimization of staffing levels to match business needs.

Features of this application include:

- Seamless integration with Mitel Networks 6110 Contact Center Management
- Skills-based Schedules
- Automated Schedules
- Customized Agent Schedules
- Accrual-based time-off planning
- Import Scheduling Information
- Employee Database
- Reporting.

Mitel Networks 6150 Multimedia Contact Center

Mitel Networks 6150 Multimedia Contact Center works with Microsoft Exchange and Outlook to provide contact centers with automatic email distribution. Emails are sent to one address, and the application manages the distribution based on agent availability and skill level. The process is similar to the way that Automatic Call Distribution works for telephone calls.

Features of this application include:

- Email auto acknowledgement
- Automatic email routing
- Customer tracking
- Multimedia Reporting
- Multimedia real-time monitoring.

Mitel Networks 6160 Intelligent Queue

Mitel Networks 6160 Intelligent Queue enhances Mitel Networks 6110 Contact Center Management. 6160 IQ is an intelligent recorded announcement device (RAD) that allows call centers to customize which messages callers will hear based on the time, date, and/or number of callers in the queue. The application is managed through a web-based interface with .WAV file recordings. The following additional options are also available:

- Static RAD
- TIQ TALK provides intelligent messaging to the caller based on real-time ACD information supplied by the 6110 CCM application
- Smart Messaging provides callers with time of day, day of week, and exception-based messaging
- Smart Choice - auto attendant feature
- Web Callback provides web users with the capability of entering their telephone number into a web page and generating a queued Callback from a contact center agent
- Voice Message Queuing provides callers the option of entering their telephone number and a voice message to generate a queued Callback from a contact center agent
- Intelligent Call Routing allows calls to be routed based on ANI, DNIS, time of day, or current queue conditions
- Call Detail Reporting.

Mitel Networks 6170 Contact Center Quality Monitoring

Mitel Networks 6170 Contact Center 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.

Call Recording features include:

- 8 - 128 channels
- Support for thousands of networked channels
- TCP/IP, NET BIOS, Token Ring, Ethernet support
- Standard 1000 hour hard disk
- Loop Playback facility
- Remote diagnostics and service
- Simultaneous record and playback from the active archive media
- Record Mode; Cascade, Sequential, or Parallel
- Compression Rate: 15,24, 32, 40, 64 Kbps selectable per channel.

Quality Monitoring 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.

Mitel Networks 6180 Contact Center Media Costing

Mitel Networks 6180 Contact Center Media Costing provides the supervisor with all the tools they require to keep track of communications costs.

- Telecom Billing System
- General Cost Allocation System
- Consolidated Reporting System
- Internet Reporting System
- Web-based Reporting.

Desktop Applications

Mitel Networks 5810 PDA Application

The Mitel Networks 5810 PDA Application increases the functionality of a Personal Digital Assistant (PDA) with the following applications:

- Telephony Features Integration (TFI) gives users the flexibility to use as their own any 5140 IP Appliance or any 5020 IP Phone that has the Mitel Networks 5423 IrDA Module
- Dial by Address Book (DBAB) allows the user to select a telephone number and dial it directly from the PDA.

Note: The Mitel Networks 5423 IrDA Module, which is available as an add-on unit for the Mitel Networks 5020 IP Phone and is integral to the Mitel Networks 5140 IP Appliance, provides the connectivity between the telephone and the PDA.

Manual Maker

Manual Maker is a tool for creating customized user guides. Using this tool, users can produce a user guide that is tailored to specified system, model, programmed features, and feature activation methods. Manual Maker also provides user guides for attendant consoles, voice mail, ACD, and subattendant features.

Manual Maker is available on CD-ROM or via the Internet.

Peripherals

Peripherals

The 3300 ICP system supports any of the following Mitel Networks peripheral devices:

- Mitel Networks 5001 IP Phone
- Mitel Networks 5005 IP Phone
- Mitel Networks 5010 IP Phone
- Mitel Networks 5020 IP Phone
- Mitel Networks 5140 IP Appliance
- Mitel Networks 5410 Programmable Key Module
- Mitel Networks 5415 Programmable Key Module
- Mitel Networks 5303 Conference Phone
- Mitel Networks 5305 IP Office Conference Unit
- Mitel Networks 5310 IP Board Room Conference Unit
- Mitel Networks 5423 IrDA Module
- Mitel Networks 5550 IP Console.

The following accessories are supported:

- Mitel Networks 3300 In-Line Power Unit
- Mitel Networks 5485 IP Paging Unit.

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

- SUPERSET 4001
- SUPERSET 4015
- SUPERSET 4025
- SUPERSET 4125
- SUPERSET 4150
- SUPERCONSOLE 1000

Analog telephones may also be connected when using the 3300 Universal ASU or 3300 ASU.

Desktop

Mitel Networks 5001 IP Phone

The Mitel Networks 5001 IP Phone is a low-cost, entry-level IP telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include:

- Three fixed-function keys
- Handset and Ringer Control
- Message Waiting Lamp
- Wall-mountable.

LL0067

Mitel Networks 5005 IP Phone

The Mitel Networks 5005 IP Phone is a low-cost IP telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include:

- Twenty-character alpha-numeric liquid crystal display (LCD)
- Twenty feature keys (6 pre-assigned)
- Two fixed-function keys
- Handset and Ringer Control
- Message Waiting Lamp
- Wall-mountable.

LL0066

Mitel Networks 5010 IP Phone

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

- Twenty-character alpha-numeric liquid crystal display (LCD)
- Seven line keys, each with a built-in line status indicator
- Six fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, and Message
- Automatic selection of prime line or ringing line
- Key selection of non-prime line
- Handset and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated Headset port.

Mitel Networks 5020 IP Phone

The Mitel Networks 5020 IP Phone is a digital telephone that connects to a 10/100BaseT Ethernet network. Features of the telephone include

- Twenty-character alpha-numeric LCD with contrast control
- Three softkeys for feature access
- Fourteen line keys, each with a built-in line status indicator
- Eight fixed-function keys: SuperKey, Cancel, Hold, Redial, Transfer/Conference, Message, Microphone, and Speaker
- Automatic selection of prime line
- Key selection of non-prime line
- Handsfree operation (half-duplex)
- Handset, speaker, and ringer volume controls (Up Arrow and Down Arrow)
- Ringer pitch control
- Message waiting lamp
- Dual Ethernet port to provide connectivity to the LAN for both your telephone and computer
- Dedicated Headset port.

LL0030

Mitel Networks 5140 IP Appliance

The Mitel Networks 5140 IP Appliance is an exciting new IP telephony offering. It combines proven, high-quality communications with superior features 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
- Standard 12 key dialpad
- Infrared transceiver lens for use with Mitel Networks 5810 PDA Application
- 6 command keys that perform context-sensitive actions shown on the LCD display
- 9 quick keys for interacting with items shown on the LCD display
- 4 navigation keys enabling you to scroll and move around pages on the display screen
- 3 telephony feature keys
- 8 applications keys for easy access to 5140 IP Appliance applications
- Ringing Indicator
- Message Waiting Indicator
- 2 10/100BaseT Ethernet ports
- Embedded Help
- Mitel Networks AC Power.

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

- Corporate Directory allows the user to scroll through names or search based on last name, full name, department, and location
- Personal Directory allows the user to maintain a list of personal contacts
- Call Log records the last 60 incoming and outgoing calls (detailing caller ID, duration, date, and time) made by and to the 5140 IP Appliance enabling users to see who called in their absence
- Bookmarks allow quick access to a customized list of URLs
- Services allow the administrator to program corporate URLs that can be accessed easily through the 5140 IP Appliance.

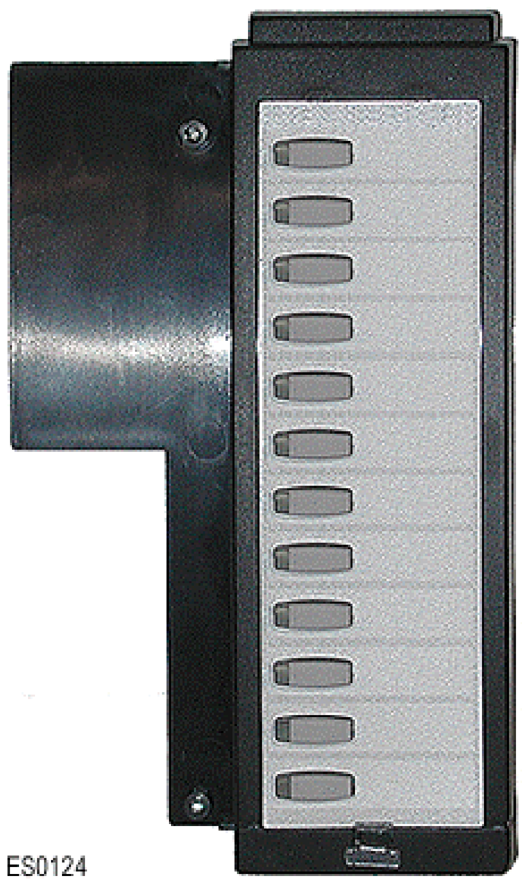
For detailed information, refer to the Mitel Networks 5140 IP Appliance help system.

LL0020

Mitel Networks 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 as a line appearance key. Each key has a Line Status Indicator that works the same way as those on the associated telephone.

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



ES0124

Mitel Networks 5415 Programmable Key Module

The Mitel Networks 5415 Programmable Key Module provides 48 additional feature keys for a 5020 IP Phone. They can be programmed as feature keys, speedcall keys, Direct Station Select keys, or as a line appearance key. 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 by using a Mitel Networks PKM Interface Module (IM). The PKM IM is installed separately at the base of telephone and is only compatible with 5020 IP Phones.

LL0035

Mitel Networks 5423 IrDA Module

The Mitel Networks 5423 IrDA (Infrared Display Adapter) Module is an optional module that attaches to a 5020 IP Phone. 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.

Mitel Networks 5303 Conference Phone

The Mitel Networks 5303 Conference Phone uses Mitel Networks acoustic beam-forming technology to produce superior performance.

Features of this unit include:

- Full Duplex operation
- 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
- 9 pre-programmed speed dials
- Time of day and date programming
- Echo cancellation
- 12 key alpha numeric keypad
- Softkeys for easy programming.

LL0046

Mitel Networks 5305 IP Office Conference Unit

The Mitel Networks 5305 IP Office Conference Unit is a high quality conference unit that uses acoustic beam forming technology to ensure superior performance. The unit is used in conjunction with the 5020 IP Phone and connects by using the telephone's headset port. This unit is designed for a private office that measures 12 feet by 15 feet (3.6 meters by 4.5 meters).

Features of the conference unit include:

- Full Duplex operation
- 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
- Power supply from a 24V wall adapter
- Simple installation
- Side Control Unit with mute, hold, and volume controls.

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

LL0027

Mitel Networks 5310 IP Board Room Conference Unit

The Mitel Networks 5310 IP Board Room Conference Unit is a high quality conference unit that uses acoustic beam forming technology to ensure superior performance. The unit is used in conjunction with the 5020 IP Phone and connects by using the telephone's headset port. This unit is designed for optimal performance in a room that measures 15 feet by 25 feet (4.5 meters by 7.6 meters).

Features of the conference unit include:

- Full Duplex operation
- 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
- Directional and Presentation Modes
- Dual color LEDs (7 on the unit in total) for visual confirmation the unit has picked up the speakers voice
- Power supply from a 24V wall adapter
- Simple installation
- Side Control Unit with mute, hold, and volume controls.

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 telephone connects to the DNI card in the peripheral cabinet. It is a single-line, digital telephone that gives users basic access to system functionality. The SUPERSET 4001 telephone has

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

ES0007

SUPERSET 4015

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

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

SUPERSET 4025

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

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

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

ES0009

SUPERSET 4125

The SUPERSET 4125 telephone connects to the DNI card in the peripheral cabinet. It is a multiline, digital telephone with

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

The telephone has the same set functionality as the SUPERSET 4025, and has a backlit digital display.

The SUPERSET 4125 supports the PKM Interface Modules for connection to additional devices.

ES0010

SUPERSET 4150

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

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

The SUPERSET 4150 also accepts PKM Interface Module that lets you connect to additional devices.

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

The SUPERCONSOLE 1000 connects to the DNI card in the peripheral cabinet.

The console has

- Fourteen hardkeys
- Four programmable firmkeys (for access to purchased options such as Hotel/Motel)
- Ten softkeys
- A dial pad (for both alphabetic and numeric input)
- Backlit display
- Volume controls
- Integral handset
- Connector for a headset
- 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 application 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 traffic 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, and Spanish.

The 5550 IP Console is sold with all the parts and 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 Millenium, 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:

- Keypad
- AC Power adapter
- Mitel Networks 4000 series handset and cord
- Handset Cradle
- Ethernet cable 10/100BaseT
- A CD-ROM containing the application software
- Designation labels for the programmable keys
- Quick Start Guide
- Installation Guide.

EE0527

Accessories

Mitel Networks 3300 In-Line Power Unit

The Mitel Networks 3300 In-Line Power Unit provides power and data communication over an Ethernet network; it eliminates the need for AC outlets and AC adapters for IP telephones. Each unit can be used to provide remote power feeding for 24 IP telephones from a centralized universal 110/220V, 60/50Hz AC input.

The 3300 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 3300 In-Line Power Unit, and the data/power output jack on the 3300 In-Line Power Unit connects to the data input jack on the IP telephone.

For more information, refer to the documentation that is shipped with the unit and the Mitel Networks 3300 In-Line Power Unit User Manual.

CAUTION: Follow all installation instructions and observe all safety precautions contained in the safety documentation provided with the 3300 In-Line Power Unit. Refer servicing to qualified service personnel only.

IP0216

Mitel Networks 5485 IP Paging Unit

The Mitel Networks 5485 IP Paging Unit is an optional module that provides paging functionality on the system. The 5485 IP Paging Unit can be a stand-alone or a wall-mounted unit. There are two LEDs that provide basic status information. The unit is powered by a 24 VDC power adapter which is supplied.

Each IP Paging Unit supports one paging zone.

For detailed information, refer to the Hardware User Guide.

IP0223

Network

Lines and Trunks

Lines

The system 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 analog 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 cabinet.
- 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 (5800m) or less on 26-gauge wire. These lines are supported by the OPS line card and are only available if a peripheral cabinet 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 (1000m) or less on 26-guage wire. These lines are supported by the DNI Line card and are only available if a peripheral cabinet 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.
- 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.
- 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 ASU or the PRI card in the DSU cabinet.
- 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 through the 3300 BRI NSU or the BRI card in the DSU cabinet.

The following analog trunks are supported:

- Analog CO trunks interface to the system by using the Loop Start (LS) ports on the 3300 Universal ASU or the Loop Start/Ground Start (LS/GS) card in the peripheral cabinet.

- E&M trunks interface to the system using the E&M trunk card in the peripheral cabinet. 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 interface to the system through the DID/Loop Tie trunk card in the peripheral cabinet.

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 for a 'cluster' of up to 80 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 that adjuncts an 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.

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.

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.

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. 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

R2 Support

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.44o 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)*	1.5197 per second 5471 per hour
Response Time Specification	Delay to Dial Tone 1 s Dial Tone Cut Off Delay 500 ms Post-Dialing Delay 1.5 s Connecting Delay 400 ms
Data Blocking Possibilities	Software <0.0001 Blocking Probability DTMF, Trunks Provisioning dependent
Note: The BHCC will vary according to individual customer configuration and usage.	

Trunking

Configuration	Trunks Echo Channels	Calls per hour From lines	Erlangs per Resource	CCS per Resource
ACD 50 Ports	3 Trunks	1350	36.8	1323
			64 Echo Channels	
ACD 100 Ports	4 Trunks	2700	73.5	2646
			128 Echo Channels	
700 IP	4 Trunks	4200	64.2	2310
			128 Echo Channels	
604 IP and 96 ONS	4 Trunks	4200	64.2	2310
			128 Echo Channels	
350 IP	2 Trunks	2100	32.1	1155

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).

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 at any station to change the COS and COR. Non-Verified Account Codes let you enter codes on the SMDR record for billing and/or call management.
Account Codes - System	System Account Codes are automatically outpulsed by the system when outgoing calls are made on a specialized carrier trunk circuit.
ACD 2000® Extended Agent Groups	This feature lets you program a maximum of 64 agent groups with up to 150 agents in each group. By using the ACD 2000 Extended Agent Groups feature package, you can assign up to 500 agents to each group; however, the maximum number of agent groups is reduced to 32.
ACD 2000 Skill-based routing	Each agent in an agent group is assigned a skill level. Calls to the group are routed to the most skilled available agent. If agents of equal skill are available, the call is routed to the longest-idle agent. To facilitate skill-based routing, agent IDs can appear in more than one agent group.
ACD Make Busy Reason Codes	ACD agents enter a reason code when telephones are put into a Make Busy state.
ACD Real Time Event	Makes the ACD Real Time Event stream a purchasable option.
Add Held	Add Held lets you move a call on Hold to another line appearance, form a conference with a call on Hold, or add a call on Hold to an existing conference.
Advanced Analog Networking	The Advanced Analog Networking (AAN) feature provides calling line identification and travelling class marks across T1/D4 trunks.
Advanced ARS	Allows day and time zones, route plans, and ARS assignment to be programmed.
Advice of Charge	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, feature configuration and administration. Each of these components offers many features not available with ANSWER PLUS - Automatic Call Distribution.
ANSWER PLUS - Mitel Networks Call Distribution	Permits the use of Recorded Announcement Devices (RADs) and a uniform call distribution to hunt groups.

Feature Name:	Description:
Attendant Access	See Attendant Directory Number.
Attendant Alarm Indications	See Attendant Console Status Display.
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 that you want other attendants to see and to store speed dial numbers that all attendants can access.
Attendant Busy-Out (Console)	Attendant Busy-Out (Console) places your attendant console in a busy-out condition (absent status) under certain circumstances. In the busy-out condition, incoming calls are automatically rerouted.
Attendant Busy-Out (Station)	Attendant Busy-Out (Station) lets you busy-out a specific station by using the attendant console.
Attendant CAS Interface	Centralized Attendant Service (CAS) interface allows 3300 ICP to be a remote node for a CAS site. CAS is an attendant call handling service that is provided at a central office switch for calls from both public and private networks.
Attendant Call Answering Priority	Attendant Call Answering Priority lets you assign priority to calls based on their destination when multiple calls are waiting, the call with the highest priority is answered first.
Attendant Call Information Display	The Attendant Call Information Display provides the attendant with information about called and calling parties.
Attendant Call Selection	Attendant Call Selection lets you choose which group of incoming calls to answer first; each group is selected by pressing a softkey on the attendant console.
Attendant Conference	Attendant Conference lets the attendant set up one or more conference connections between central office trunks and internal stations.
Attendant Consoles	See Peripherals.
Attendant Consoles (Multiple)	Multiple Attendant Consoles can be supported.
Attendant Console Firmkeys	Attendant Console Firmkeys on your console can be programmed as one of the following feature keys: Phonebook, Guest Service (Hotel/Motel), Trunk Status, Alarm, SMDA, Select Option, or blank (no application).
Attendant Console Status Display	Attendant Console Status Display on each attendant console displays various parameters such as Day/Night Service, Attendant Status, and Alarm Status.
Attendant Directory Number	Attendant Directory Number lets you dial an attendant directory number (typically "0") to reach the attendant. Separate directory numbers can be programmed for each attendant console.
Attendant Help	Attendant Help provides online assistance.
Attendant Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features.
Attendant Identity Information Display	Attendant Identity Information Display lets you view the console's prime directory number, the PB software version, and the console's hold slot number.
Attendant Language Selection	Enables attendant to choose the language of operation for the attendant console (English, French, German, or Italian).

Feature Name:	Description:
Attendant Lockout	Attendant Lockout prevents the attendant from re-entering a call once the attendant has released.
Attendant Messaging	Attendant Messaging lets you activate a message waiting condition on a station from the attendant console. The condition can be queried or canceled by the attendant or by a station user with the appropriate Class of Service.
Attendant Metered Calls	Attendant Metered Calls lets you use the attendant console to track the cost of outgoing trunk calls.
Attendant New Call Tone	Attendant New Call Tone notifies you of new calls to the attendant console through an audible indication.
Attendant Position Busy-Out	See Attendant Busy-Out (Console).
Attendant Recall	Attendant Recall automatically alerts the attendant when a trunk call has been extended to an idle station and not answered within a specified time-out period or when a call on Hold at the console has not been answered within a selected time.
Attendant Ringer Control	Attendant Ringer Control lets you mute the attendant console ringer. When the attendant console ringer is muted, incoming calls continue to be indicated by the Call Waiting prompt at the top of the display.
Attendant Scratch Pad	The Attendant Scratch Pad is your own 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	Attendant Serial Call automatically returns a call to the attendant console when the caller finishes with the called party.
Attendant Setup and Cancellation of Station Features	The attendant can setup and cancel certain station features such as Call Forward, Do Not Disturb, Callback, and Reminder.
Attendant System Login	The attendant has access to some programming functions from the attendant console. To access these programming functions, the attendant must log on.
Attendant Tone Signaling	Attendant Tone Signaling lets the attendant send tones over the circuit once a call has been established.
Attendant Trunk Group Busy Status	Attendant Trunk Busy Status Display lets you display and/or print the busy status of the system trunk groups from the attendant console.
Auto-Answer	Auto-Answer lets you automatically answer calls that ring your Prime line.
Auto-Hold	Auto-Hold lets you automatically place an active call on Hold when you press a line key to originate or receive another call.
Automatic Route Selection (ARS)	Automatic Route Selection (ARS) simplifies local and long distance dialing by automatically selecting the most convenient and cost-effective route and by inserting and/or deleting the proper routing digits.
Autovon	Allows the system to connect to with Autovon networks (defense switched networks and Canadian switched networks) for incoming and outgoing calls.

Feature Name:	Description:
BRI (Basic Rate Interface)	BRI is a basic ISDN service consisting of two 64Kbps channels and a single 16Kbps channel. This feature is supported on the 3300 ICP with the 3300 BRI Network Services Unit.
Broadcast Groups	See Groups - Key System and Multicall.
Broker's Call	Broker's Call lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.
Busy Dial Through	Busy Dial Through lets you dial a feature access code sequence when a busy condition is encountered. See Callback and Camp-on.
Busy Override	See Override.
Calibrated Flash	See Flash - Calibrated.
Callback	Callback lets you request that the system notify you when a busy line becomes idle or when an unanswered station goes off-hook and on-hook.
Callback – System Programmable	Callback - System Programmable lets you program the destination of a matured callback set against a key line or multicall line group.
Call By Name	See Phonebook.
Call Coverage	Call Coverage is provided through a combination of features: Call Rerouting, Call Forward, Do Not Disturb, and Answer Plus - Mitel Networks Call Distribution.
Call Duration Display	Call Duration Display provides you with a display of the call duration for incoming and outgoing calls in one minute increments (starting at 0:00) from the beginning of the call to the end of the call.
Call Forward	Call Forward lets you redirect incoming calls to an alternate number.
Call Forward - Cancel All	Call Forward - Cancel All lets you cancel all types of Call Forward.
Call Forward - Follow Me - End Chaining	Call Forward - Follow Me - End Chaining ensures that calls are not further redirected.
Call Forward - Follow Me - Reroute When Busy	Call Forward - Follow Me - Reroute When Busy forwards the call to the original set's First Alternative Rerouting if the call forward destination is busy.
Call Forward - Forced	Call Forward - Forced lets you manually redirect an incoming call on your Prime or private line to another number.
Call Forward - Override	Call Forward - Override lets you bypass any Call Forward condition that is set at the station that you are calling.
Call Hold	See Hold.
Call Park	Call Park lets the attendant Hold a call so that a telephone user can remotely retrieve the call.
Call Pickup	Call Pickup lets you answer an incoming call that is ringing at another station.
Call Privacy	Call Privacy protects a call from audible Call Waiting tones, as the result of a camp-on, and prevents intrusion of any kind (for example busy override).

Feature Name:	Description:
Call Release	See Release.
Call Rerouting	Call Rerouting lets the system redirect calls to alternate answering points or devices under specified conditions. Call Rerouting may be used to redirect calls always (in Day, Night 1, and/or Night 2 mode) or under busy, no answer, or Do Not Disturb conditions.
Call Split	See Conference Split.
Call Swap	See Swap.
Call Transfer	See Transfer.
Call Waiting Swap	Call Waiting Swap lets you use the switch hook to alternate between two calls when a party is in Call Waiting for your station or when you have a call on Consultation Hold.
Camp-on (Call Waiting)	Camp-on, or Call Waiting, lets you notify a busy party that you are waiting. An attendant may also put a call through to a busy station to indicate they are waiting. Upon hearing the Call Waiting tone, the busy party can either respond or finish the current call.
Camp-on Tone Security	Camp-on Tone Security prevents you from hearing Camp-on tone. If any party in a call has this option enabled, no Camp-on tone is returned to anyone in the call.
Centralized Attendant Service (CAS) interface	See Attendant CAS Interface
Class of Restriction	Class of Restriction (COR) limits a station's access to specified numbers. A station may have three CORs (Day/Night1/Night2 service), and the COR may also be changed by using a Verified Account Code.
Class of Service	Class of Service (COS) defines a station or trunk's feature and timer options. A station or trunk may have three COSs (Day/Night1/Night2 service), and the COS may also be changed by using a Verified Account Code.
Clear All Features	Clear All Features lets you cancel most of the features activated on your extension or another user's extension.
Compression	Compression allows IP calls to be made utilizing less bandwidth than an uncompressed call.
Conference	Conference lets you connect three or more people into a single telephone conversation. While you are in a Conference, you can use any of the features that would normally be available during a two-party call.
Conference Split	Conference Split lets you separate a 3-party conference so that you can speak privately with one of the parties. While you are speaking privately with one party, the other party is on Consultation Hold.
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	The date and time is set through the System Administration Tool. This data appears on all Station Message Detail Recording (SMDR), traffic measurements, data dumps, display telephones, and attendant consoles.

Feature Name:	Description:
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 dial tone (also called interrupted dial tone) 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	Dialed Number Editing lets you edit numbers during dialing.
Dialing - Conflicting Numbers	The system can differentiate between conflicting numbers such as 1-0-0-0-0 and 1-0-0-0. In this example, if the 5th digit is not dialed within a time-out period, the system assumes that the dialed sequence is complete and makes the call.
Direct-In Lines (DIL)	Direct-In Lines (DIL) allow incoming trunks to be assigned to a specific station or hunt group so that calls from the trunk ring the station or hunt group directly.
Direct Inward Dialing (DID)	Direct Inward Dialing (DID) allows incoming calls on designated trunks to directly access predefined stations (or other answering points) on the system.
Direct Inward System Access (DISA)	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)	Direct Outward Dialing (DOD) lets you make external calls without the assistance of the attendant.
Direct Page	Direct Page allows you to page another telephone over its built-in speaker. See Off-Hook Voice Announce.
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	Display Contrast Control lets you adjust the contrast of the alphanumeric display.
DNI	Allows the programming of Mitel Networks digital devices
Do Not Disturb	Do Not Disturb (DND) lets you place your set in an apparent busy condition without affecting the outgoing functionality. If someone calls your set while DND is activated, he or she will hear special busy tone.
DTMF Keypad Support	DTMF Keypad Support lets ONS/OPS extensions use all 16 keys on a 4x4 DTMF keypad. The additional row of four keys (ABCD) is used to access features in the system.

Feature Name:	Description:
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	Feature Keys let you activate features without dialing feature access codes.
Flash - Calibrated	Flash - Calibrated provides an alternative method of generating a Switchhook Flash.
Flash - Switchhook	Flash - Switchhook lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features.
Flash - Trunk	Flash - Trunk lets you single or double flash a trunk in order to access Centrex™ features.
Flexible Answer Point	Flexible Answer Point lets station and console users program a night answer point for their incoming trunk calls.
Ground Button	A Ground Button (Recall Button) lets you place a call on Consultation Hold and return to dial tone so that you can invoke station features. The Ground Button provides an alternative method of producing a Switchhook Flash.
Group Page	Group Page lets you page a group of telephones over their built-in speakers.
Groups - Key System and Multicall	Key System Groups and Multicall Groups let multiple telephones share the same extension number. Incoming calls ring all of the idle stations, and the stations stop ringing when one member answers the call.
Handset Receiver Volume Control	Handset Receiver Volume Control lets you adjust the volume of the handset receiver.
Handsfree Operation	Handsfree Operation lets you use your telephone without lifting the handset.
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	Headset Operation lets you use a Headset to make and receive telephone calls.
Hold	Hold lets you temporarily suspend a telephone call. While the call is on Hold, you can use the other telephone features. The call can be either retrieved at the originating telephone or another telephone.
Hold on Hold	This feature allows both parties of a two-party call to put the call on Hold.
Hotel/Motel	Provides features commonly used by hotels, motels, hospitals, as well as a Property Management Interface.

Feature Name:	Description:
Hotline	Hotline limits your access to a designated answer point. The system automatically dials the answer point when you go off-hook. The designated answer point can be another station, an attendant, a trunk, or a hunt group.
Hunt Groups	Hunt Groups let you dial a pilot number and have the call completed to the first idle station in a group of stations. Any station within a Hunt Group may be accessed directly by dialing the station number.
Intercept Handling	Intercept Handling lets the system control what happens to a call when the call cannot be completed to the required destination. A call may be routed to a tone or to a directory number. Two alternate destinations may be programmed for each condition.
Interconnect Restrictions	Each peripheral device is assigned an Interconnect Number that is used to restrict one device from connecting with another. Interconnect Restrictions can be used to restrict access to certain trunks, stations, or equipment (i.e. data communications equipment). The restriction is also a function of the direction of the call.
IP Networking	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.
Language Change	Language Change lets you change the language of the telephone softkeys and prompts to any one of the following languages: English, French, Italian, German, Spanish, or Dutch.
Line Types and Appearances	Line appearance keys are single or shared lines that appear on the telephone programmable keys. There are three types of lines: Prime, Non-Prime, and No Where Prime.
Line Appearance Ring Types	Each line appearances can be programmed to ring in a different manner.
Maintenance	The system provides extensive maintenance coverage. All types of peripheral hardware are periodically tested by the system. Maintenance users may also test individual circuits on demand.
Meet Me Answer	Meet Me Answer lets a paged party respond to a Group Page even if they do not know the identity or location of the paging party.
Messaging - Advisory	Messaging - Advisory lets you select a short advisory message to show display set users who call your telephone.
Messaging - Callback	Callback Messaging lets you leave a callback message on a telephone when the called party is busy or does not answer. When you receive a callback message, you can review the message on the display (if applicable) and/or call the sender back.
Messaging - Dialed	Dialed Messaging lets you leave a message-waiting indication on a telephone. When you receive a message-waiting indication, you call your message taker to accept the message.
MNMS	Supports OPS Manager functions.
MSDN/DPNSS	MSDN/DPNSS is a digital signaling system that provides many other features and used within a private network of PBXs.

Feature Name:	**Description:**
MSDN Release Link Trunk	This feature allows the attendant to make a call by using the same incoming trunk. It helps to provide centralized attendant service by allowing the 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 which are associated with the mailboxes of other stations.
Music	Music lets you listen to the Music On Hold music source through the speaker of the telephone.
Music On Hold	Music On Hold provides callers with music while they are waiting for a call to be completed. Music On Hold is provided when a call is on Hold, when a call is transferred to a busy party, or when a call is in Call Waiting for a station. The customer provides the music source.
Networking	The system supports both analog and digital networking. See Node ID Recognition and Uniform Numbering Plan.
Networked ACD	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 first entered the network.
Networked Group Page	Group Paging can be completed across a network or cluster. This allows a set on system A to page a specific group defined on system B.
NI3 Calling Name Delivery	NI3 Calling Name Delivery allows the called party to see the name of the caller on the telephone display screen if the caller has programmed Calling Name to "Allow" through IMAT. NI3 supports both incoming and outgoing calls for the system T1 card and is supported by the 3300 Universal NSU.
Night Service	Night Service lets you redirect calls to alternate answer points for individual trunks. The answer point used depends on the selected mode of operation (Day, Night 1, or Night 2).
Night Service - Automatic	Automatic Night Service places the system into Night service automatically if all attendant consoles are unable to receive calls or if all attendant consoles are inactive and the time-out period has expired.
Node ID Recognition	Node ID Recognition lets a system in a network determine if an incoming call applies to it or to another system in the network.
Non-Busy Station	Non-Busy Station lets you program an extension never to return busy tone. This feature is used for special situations such as emergency stations.
Non-DID Extension	Non-DID Extension allows the system to support sets that are not directly accessible to DID trunks. These calls are transferred to Non-DID Extensions by an Intercept Handling point (such as an attendant or a station).

Feature Name:	Description:
Off-Hook Voice Announce	Off-Hook Voice Announce lets you receive a Direct Page during a handset or headset call. See Direct Page.
Overlap Outpulsing	Overlap Outpulsing reduces post-dialing delay when trunk calls are originated. Once a route has been determined by ARS, a trunk is seized and dial pulses or tones are outpulsed to the CO. These outpulses are sent before the user has finished dialing to allow faster call setup on analog trunks.
Override	Override lets you enter a conversation at a busy station or ring a station with Do Not Disturb activated. Before you enter the conversation, all parties receive a warning tone.
Override Security	Override Security prevents users from using Override on your station.
Paging	Paging lets you connect to loudspeaker/paging equipment to access individual paging zones or all paging zones simultaneously. Before you are connected to the paging equipment, you will hear a two-second burst of tone.
Phonebook	Phonebook lets you locate and telephone a system user based on his or her name, extension number, department, and/or location.
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 both system printers for its own applications (such as SMDR and maintenance) and dedicated data communications printers.
Priority Queuing	Priority Queuing ensures that calls are handled in order of priority. When internal or external callers must wait for calls to be completed, they are placed into a queue and assigned an access priority.
Privacy Release	Privacy between users who share line appearances in key systems groups is automatic. The privacy release feature allows users to release privacy during a call to allow another member of the key system group to intrude on the call.
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 systems Universal Network Services Unit.
Recall	Recall lets an incoming caller, who has been transferred to an idle station and not answered within a specified time-out period, call back the last party who handled the call. Similar time-out Recalls occur for parties who were transferred to busy stations or who were placed on Hold.
Recall Button	See Ground Button.
Redial	Redial lets you automatically dial the last number that you manually dialed.

Feature Name:	Description:
Redial - Saved Number	Redial - Saved Number lets you save a number for future dialing. The number remains saved until a replacement number is saved.
Release	Release lets you forcibly release from an attempted connection to an external party without going on-hook. Release is useful when you encounter a busy or unavailable external party that you are attempting to add to a Conference.
Reminder	Reminder lets you program your set to ring and provide a message at a specified time within the 24-hour period.
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	Ringer Control lets you adjust the volume and pitch of the telephone ringer.
Ringing - Discriminating	Discriminating Ringing lets you distinguish between incoming internal calls, incoming trunk calls, tie line calls, and Callbacks by using different ringing patterns (cadences).
Ringing - Discriminating (Optional)	Optional Discriminating Ringing lets you change the Discriminating Ringing patterns on ONS/OPS lines so that you hear internal ringing (1 second on and 3 seconds off) for both internal and external calls.
Ringing Line Select	Ringing Line Select lets you answer any ringing line by going off-hook.
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.
Speech Recognition Softkey Support	Provides quick and easy access to the Speech Recognition voice recognition system.
Speaker Volume Control	Speaker Volume Control lets you adjust the volume of the telephone speaker.
Speed Call Keys	Speed Call Keys let you store and dial frequently used numbers by using the personal keys on your telephone.
Speed Call - Pause	When the system encounters a Pause while dialing a Speed Call string, the system ceases dialing for the duration of the Pause. When the Pause ends, dialing resumes.
Speed Call - Personal	Personal Speed Calls let you store and dial frequently used numbers by using access codes and index numbers.
Speed Call - System	System Speed Call lets you dial stored system numbers.
Speed Dial	See Speed Call.
Station Message Detailed Accounting (SMDA)	Station Message Detailed Accounting (SMDA) lets the system accumulate meter pulses (up to an assigned buffer size) that can be read, printed, and cleared from a console. You can collect meter pulses by using either a device (device meter unit accumulation) or an account code (account code meter unit accumulation).
Station-To-Station Dialing	Station-To-Station Dialing lets you dial any other station directly.

Feature Name:	Description:
Suite Service	Suite Service provides the ability to group a number of telephone lines through interconnected hotel/motel rooms, or suites, for the purposes of billing and shared telephone service.
Swap	Swap lets you temporarily suspend a telephone call while you originate a new call. Once the new call has been established, you can alternate between the two calls.
Switchhook Flash	See Flash - Switchhook.
System Access Authorization	Administrative access to the system is controlled by passwords. Different passwords are assigned for each of the five levels of access.
System Alarm Indications	See Alarms and Attendant Console Status Display.
System Fail Transfer	See System Fail Transfer.
T1/D4	Provides support for T1 Channel Associated Signaling. A Dual T1 card is required.
TAPI Support	Supports MiTAI and TALK TO TAPI computer telephony interfaces.
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' telephones.
Tie Trunk Support	Tie trunks terminate on the attendant console, at station sets, in hunt groups, or on night bells. They may also be arranged as dial-in tie trunks or tandem trunks. Like CO trunks, tie trunks are arranged in groups.
Timed Reminder	See Reminder.
Toll Control	Toll control allows or denies access to specified routes, CO exchanges, and directory numbers.
Tone Demonstration	Tone Demonstration lets you hear the tones provided on the 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 based on system usage to allow better system resource management.
Transfer	Transfer lets you move a call from one telephone to another. Before completing a Transfer, you can consult privately with the third party and swap between private conversations with each of the parties.
Transmission Tests	Transmission Tests let you perform the following tests on a trunk: milliwatt test, balance test, and 100 test.
Trunk Access	Trunk Access lets you access a specific trunk directly. No toll control or ARS checking is done when you use Trunk Access. This feature is used when a maintenance telephone is required

Feature Name:	**Description:**
Trunk Answer From Any Station (TAFAS)	Trunk Answer From Any Station (TAFAS) lets you answer any call that rings a night bell. Once you answer the call, you can use any of the features that are normally available at the station.
Trunk Busy-out	Trunk Busy-Out lets you busy-out a specific trunk. When you perform a Trunk Busy-Out, the trunk is busied out if it is idle; if the trunk is in use, it is busied out as soon as it becomes idle. When you busy-out the trunk, it cannot be accessed.
Trunk Group Busy Status	Enables attendants to query the status of trunk groups from the attendant console.
Trunk Group Hunting	Trunk Group Hunting lets you search for trunk groups in either a terminal or circular pattern. In a terminal hunt group, trunks are always selected in a predetermined order. In a circular hunt group, trunks are selected in a distributed manner (the first free trunk after the last one used becomes the new first choice).
Trunk Labels	Trunk Labels may be assigned to individual trunks or groups of trunks. When a trunk call appears at an attendant console, the trunk label and trunk number are displayed.
Trunk Select - Direct	Direct Trunk Select lets you access an outside trunk for the purposes of originating and receiving external calls. Because the trunk is assigned to a line appearance, you can access the trunk to make or answer calls without the need for trunk access codes.
Trunk Support	The 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 different system configurations.
Voice Mail	The system has its own integral voice mail system that supports up to 750 voice mail-boxes.
Voice Mail Interfaces	Most voice processing systems work in conjunction with the system. The system provides the following voice processor interfaces: Voice Mail - Digital E&M Interface Voice Mail - ONS Interface Voice Mail – Integral
Voice Mail Softkeys	Provides the user with a quick and convenient method to access voice mail. Access to the system is provided through context sensitive softkeys presented on the IP telephone.
XNET	Proprietary switched MSDN/DPNSS networking over the PSTN.

Solutions

Solution Overview

The purpose of this section is to provide examples of different customer requirements and a possible solution that could be provided by the 3300 ICP and Mitel Networks applications.

- Standalone Site
- Multiple System Site
- Installed Base

Standalone Site

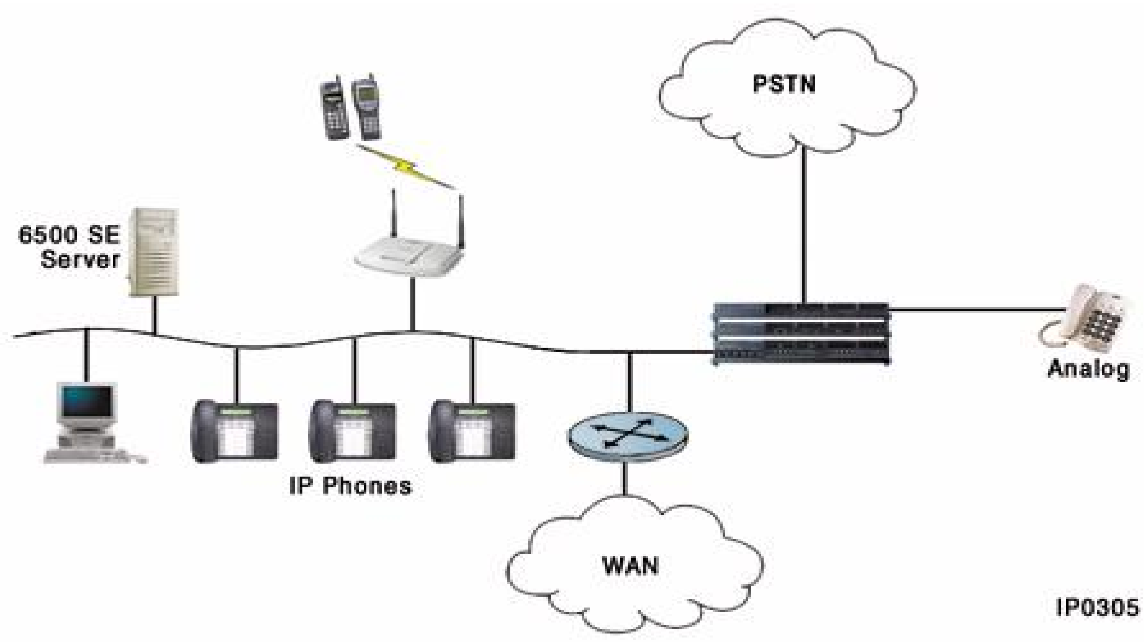
The Customer

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: manufacturing, marketing, and sales are all located in the same building.

The Problem

As with any large manufacturing operation, efficiency and cost containment are essential. There were several areas of concern to RGA's management. First, sales people complained that they were having difficulty prioritizing the volume of voice messages that they were receiving, resulting in customer complaints about response time. Second, troubleshooting the production lines was becoming more expensive as production volume and complexity increased. Third, customer service was having trouble keeping up with orders and support calls to satisfy the demands of this growing business.

The Solution



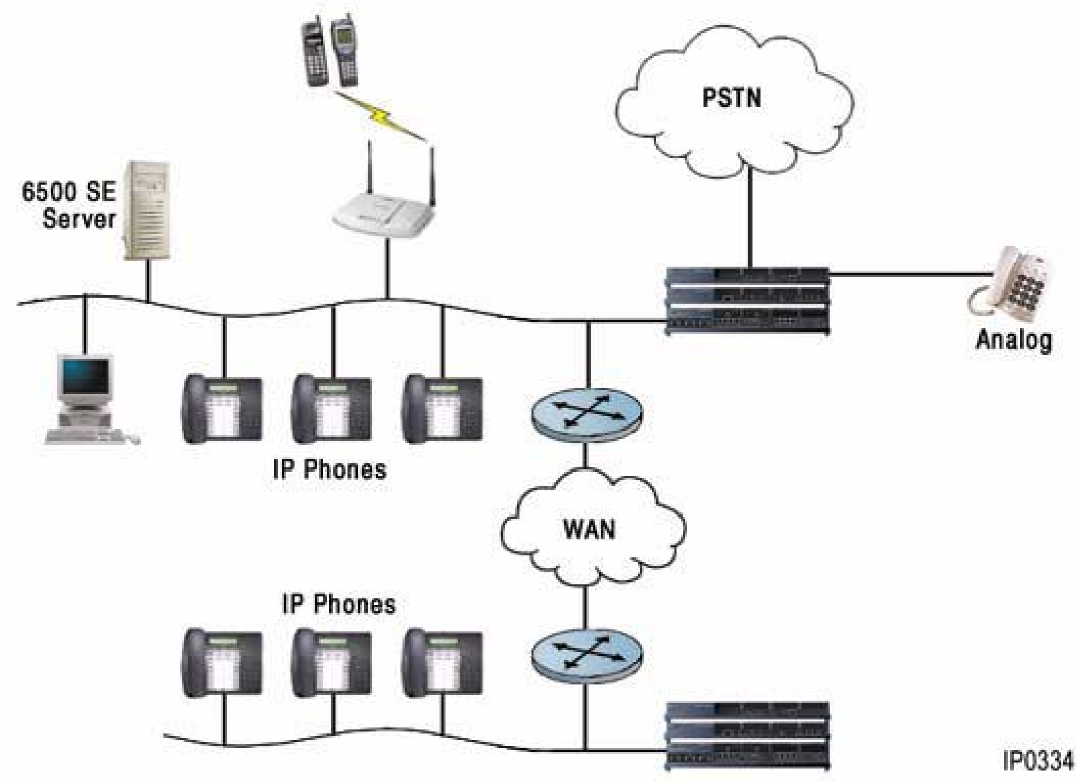
Management teamed with a Mitel Networks VAR to evaluate and implement an integrated communications solution that would meet their needs today and into the future. The foundation of the solution was the Mitel Networks 3300 Integrated Communications Platform (ICP). The 3300 ICP provides voice communications equal or superior to today's best PBXs all based on leading edge Voice-over-IP (VoIP) technology. The 3300 ICP supports all the call features and reliability upon which Mitel Networks has built its reputation for almost 30 years and supports many new features and applications as well.

The Mitel Networks 6500 Speech-Enabled Unified Messaging solution was selected to allow the sales team to efficiently manage messages (voice, e-mail, and fax), ensuring customer satisfaction. With a single message store, the 6500 SE Unified Messaging uses natural speech navigation of the inbox for e-mail, voice mail, and fax. This provides the user with the flexibility to manage messages based on sender, date, or type. It further allows the user to forward or reply to the message with voice or to simply return the call without having to look up the number through tight integration with Microsoft Exchange. To further enhance productivity, users can check their calendar, make appointments and meeting requests and create tasks all through the voice user interface. Being able to manage messages like this while on the road allows sales people to keep in touch and close more business while on the road.

Next it was decided to provide in-building mobility to technicians responsible for troubleshooting assembly lines and desktops. The Mitel Networks 3300 ICP supports Symbol Spectrum24 and Netvision allowing users full mobility while keeping in touch. Being able to make and receive calls from anywhere in their facility allows technicians to consult with colleagues about a problem as they are working on resolving it. This ensures efficient problem resolution saving a technician hours per week. And because the system is IP based, technicians can easily check the trouble ticket database to ensure the next ticket they solve is a high priority so that their time is spent on problems with the highest business value.

The integrated ACD functionality of the 3300 ICP combined with the Mitel Networks 6100 Contact Center Solutions (CCS) applications provides management with the tools required to efficiently manage their contact center. 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 their web browser, supervisors can remove ACD queues from service in times of low call volume, and return them to service when the inbound call traffic increases. Using the Mitel Networks 6120 Contact Center Scheduling solution, management can create customized schedules for agents ensuring the right amount of staff to maintain 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 users



Multiple Site

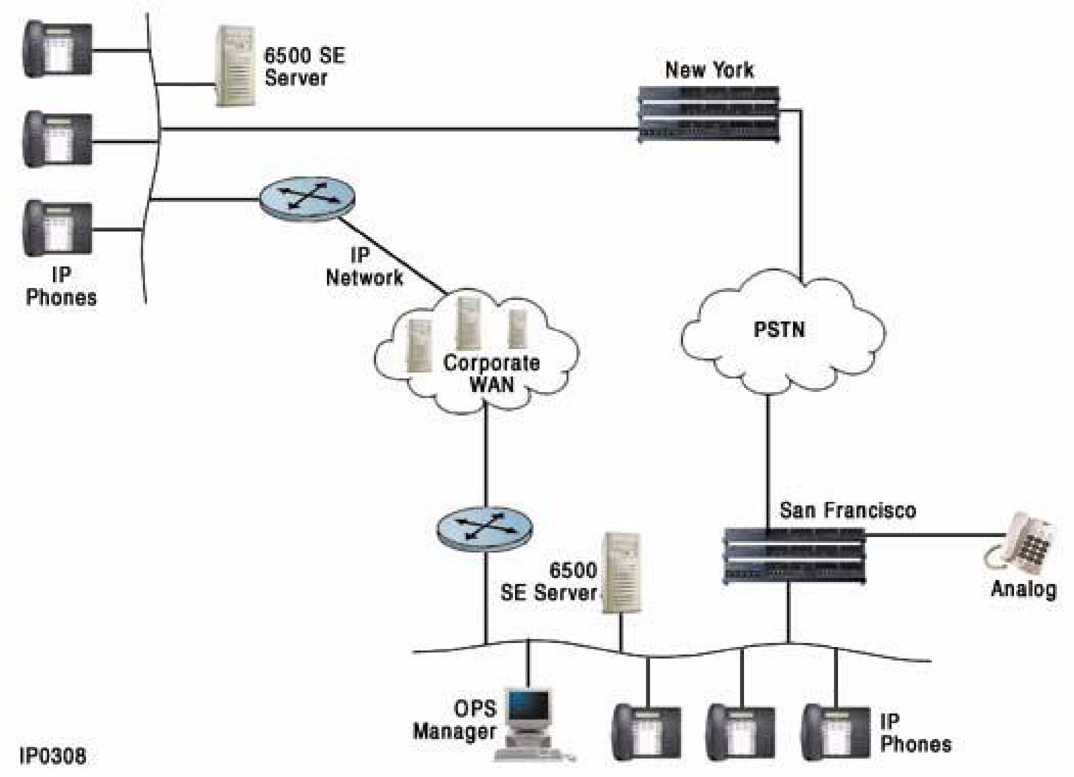
The Customer

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

The Problem

Excellent Advertising (EA) is concerned that their cost of doing business is getting out of control. Their long distance bills are spiraling upwards, and managing two independent PBXs is becoming increasingly difficult. Employees complain about wasted time dialing network access codes to call their colleagues on the other side of the country. The sales team cannot keep up with messages left by clients as they travel developing new business. The managing partners have decided that their legacy telecommunications equipment is starting to impact their ability to be successful. It was time for a change.

The Solution



Central to the solution that EA's communications service provider recommended was the Mitel Networks 3300 ICP. The 3300 ICP supports flexible networking as well as natural speech recognition technology, which deliver tangible business advantages.

EA's solution consists of two 3300 ICPs, one located in New York and one in San Francisco. The two 3300 ICPs were connected using the integrated IP Networking feature of the 3300 ICP, thus leveraging the corporate WAN. By sending voice traffic over the corporate WAN, significant long distance savings were realized.

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

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

OPS Manager was installed in New York to 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. Managers have centralized control over all elements, anytime and from anywhere.

Mitel Networks 6500 Speech-Enabled 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 allows sales staff to immediately reply to an urgent message from a valued client instead of having to answer all messages sequentially in the order that they arrived. This results in superior customer service and client satisfaction.

Installed Base Migration

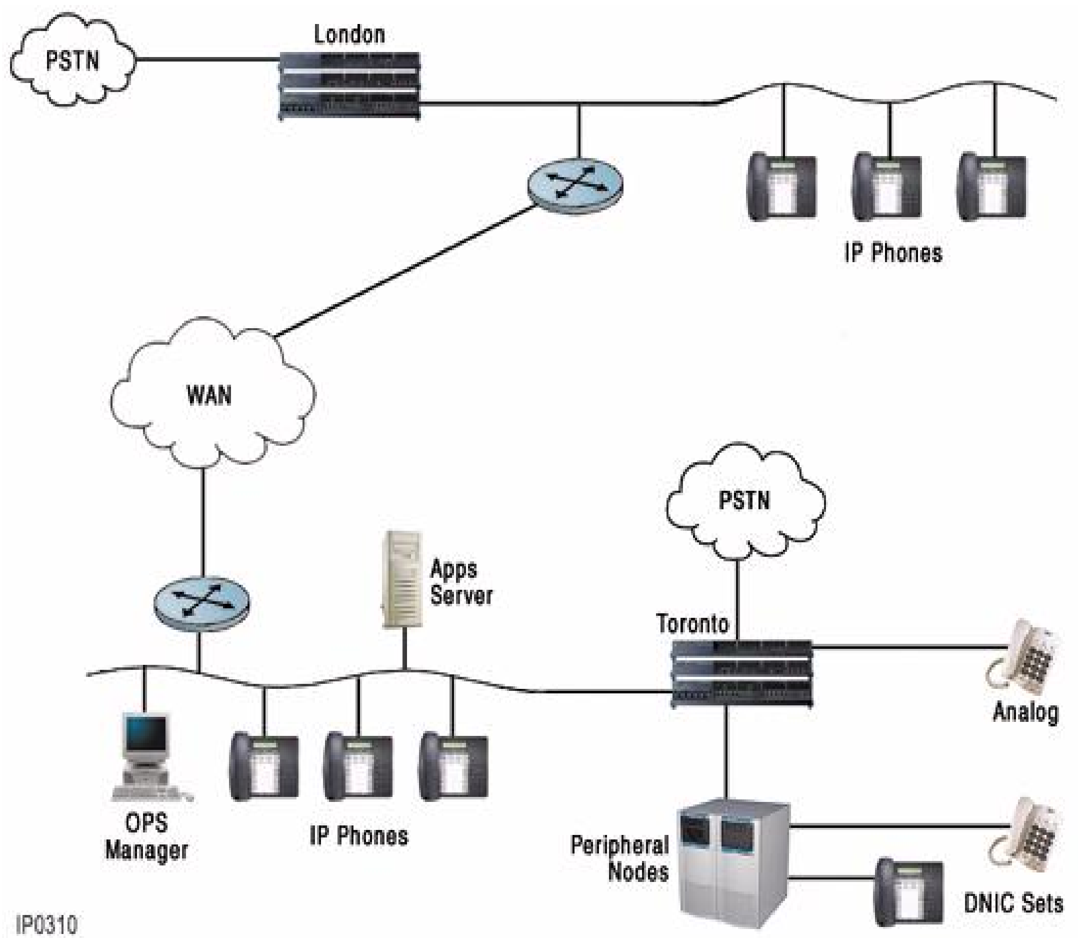
The Customer

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 5 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. Most of their experienced staff will remain in Canada and so they expect a high volume of overseas long distance calling. RGRE currently owns a Mitel Networks SX-2000 and wants to ensure that they maintain and future proof their investment as they expand .

The Solution



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 since 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 decided to route all long distance traffic over the corporate WAN. This was easily accommodated by the 3300 ICP's integrated IP Networking feature. A 3300 ICP was installed in Toronto and was connected to the 3300 ICP in London via IP Networking. In addition to realizing significant cost savings, RGRE was 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 TDM phones, providing investment protection for the existing equipment. As the 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. Managers have centralized control over all elements, anytime and from anywhere.

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