
Nortel Communication Server 1000

Nortel Communication Server 1000 Release 4.5

Communication Server 1000S

Overview

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Revision history

August 2005

Standard 3.00. This document is up-issued to support Communication Server 1000 Release 4.5.

September 2004

Standard 2.00. This document is up-issued for Communication Server 1000 Release 4.0.

October 2003

Standard 1.00. This document is a new NTP for Succession 3.0. It was created to support a restructuring of the Documentation Library. This document contains information previously contained in the following legacy document, now retired: System Overview (553-3023-010).

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About this document

This document is a global document. Contact your system supplier or your Nortel representative to verify that the hardware and software described are supported in your area.

Subject



WARNING

Before a CS 1000S system can be installed, a network assessment **must** be performed and the network must be VoIP-ready.

If the minimum VoIP network requirements are not met, the system will not operate properly.

For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Data Networking for Voice over IP* (553-3001-160).

This document provides an overview of the general design and features of the CS 1000S system.

Note on legacy products and releases

This NTP contains information about systems, components, and features that are compatible with Nortel Communication Server 1000 Release 4.5 software. For more information on legacy products and releases, click the

Technical Documentation link under **Support & Training** on the Nortel home page:

www.nortel.com

Applicable systems

This document applies to the Communication Server 1000S (CS 1000S) system.

Intended audience

This document is intended for individuals responsible for the sale, acquisition, planning, or installation of a CS 1000S system.

Conventions

In this document, the CS 1000S system is referred to generically as “system.”

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Data Networking for Voice over IP* (553-3001-160)
- *IP Peer Networking: Installation and Configuration* (553-3001-213)
- *Branch Office: Installation and Configuration* (553-3001-214)
- *WLAN IP Telephony: Installation and Configuration* (553-3001-304)
- *Integrated Conference Bridge: Service Implementation Guide* (553-3001-358)
- *Hospitality Integrated Voice Services: Service Implementation Guide* (553-3001-359)
- *Integrated Recorded Announcer: Service Implementation Guide* (553-3001-360)

- *Integrated Call Director: Service Implementation Guide* (553-3001-361)
- *Integrated Call Assistant: Service Implementation Guide* (553-3001-362)
- *IP Line: Description, Installation, and Operation* (553-3001-365)
- *Telephones and Consoles: Description, Installation, and Operation* (553-3001-367)
- *IP Phones: Description, Installation, and Operation* (553-3001-368)
- *DECT: Description, Planning, Installation, and Operation* (553-3001-370)

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Description and features

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Introduction



WARNING

Before a CS 1000S system can be installed, a network assessment **must** be performed and the network must be VoIP-ready.

If the minimum VoIP network requirements are not met, the system will not operate properly.

For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Data Networking for Voice over IP* (553-3001-160).

The CS 1000S is an IP PBX that was introduced to address the increasing demand for Voice over IP (VoIP) and support the convergence of voice and data networks. In addition to supporting powerful new IP features, including Session Initiation Protocol (SIP), the CS 1000S provides continued support of the traditional Meridian suite of features.

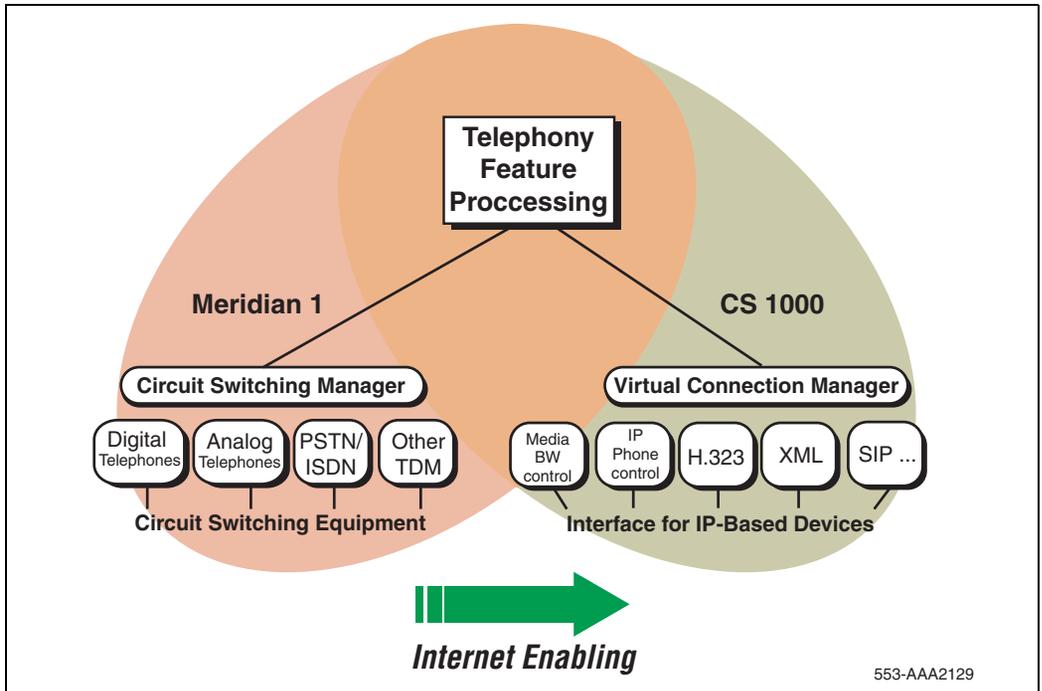
CS 1000S expands the system's capabilities to leverage the flexibility of IP WANs. It provides seamless network integration, simplified management, greater flexibility in network deployment, and reduced costs for supporting an increasingly distributed global user community.

CS 1000S is more than an IP PBX. It is the architecture that evolves, using open standards, to deliver future Advanced Applications, Interactive Multimedia Services, Integrated Desktops, and Management Services.

Migrating to an IP telephony network

CS 1000S architecture provides an IP-based alternative to traditional Meridian 1 architecture. It is designed to enable IP-based communication models while protecting the investment of Meridian 1 end users as they migrate to new technologies. Figure 1 illustrates the key components of this architecture.

Figure 1
Software architecture for telephony



Telephony Feature Processing is based on the existing Meridian 1 Call Processing software and provides industry-leading feature capabilities.

The Virtual Connection Manager allows IP-based devices that are registered with the Call Server to access the same feature capabilities as existing telephony devices. In this way, the Meridian 1 feature set is made available to evolving IP devices and soft clients, and is the switching foundation of CS 1000S. In addition to access to the Telephony Feature Processing of a Call Server, devices can also access other services in the network, such as XML and IM. The Virtual Connection Manager supports IP Phones and a wider suite of devices and services.

The Circuit Switching Manager evolved from Meridian 1 software. This migration strategy enables Telephony Feature Processing for future releases of CS 1000S to control existing Meridian 1 hardware components. In this way, the investment in existing equipment is preserved while new capabilities are added at a managed pace.

Features

Key features of the CS 1000S system are as follows:

- capacity
- flexibility and redundancy
- software applications
- enhanced system management
- terminal equipment
- interworking/interoperability

Capacity

The CS 1000S Call Server can support up to four Media Gateway 1000S (MG 1000S) systems, and each MG 1000S can support a Media Gateway 1000S Expander (MG 1000S Expander).

The MG 1000S has a Small System Controller (SSC) card and has four available slots for flexible configurations of line and trunk cards.

The Call Server can support up to 1000 IP Phones.

Flexibility and redundancy

The CS 1000S can be deployed in many flexible configurations in a LAN or WAN environment. The system can be deployed in a single location, multiple buildings in a campus setting, or a network of multiple locations with distributed Call Servers or branch offices.

The CS 1000S provides a number of capabilities to ensure that telephony is available in the event of equipment failure at a site. CS 1000S provides the following component redundancy:

- Call Server with automatic database distribution
- Signaling Server software, including H.323 Gateway and IP Phone software
- H.323 Gatekeeper
- H.323 Gateway interface to Gatekeeper
- Campus-distributed MG 1000S in Survival Mode

Refer to “Configuration options” on [page 33](#) for more information on system deployment and reliability strategies.

Software applications

CS 1000S supports a broad suite of applications, including the following:

- CallPilot applications
- Symposium Call Center Server
- Nortel Integrated Recorded Announcer
- Nortel Remote Gateway 9150
- Nortel Integrated Call Assistant
- Nortel Integrated Conference Bridge
- Nortel Integrated Call Director
- Nortel Hospitality Integrated Voice Services

CallPilot applications

CallPilot is a multimedia messaging system that offers a single solution for managing information, including the following:

- voice, fax, and e-mail messages
- telephone calls (including conference)
- calendars

- directories
- call logs

CallPilot 2.0

CallPilot 2.0 includes the 501t and the 1002rp messaging platforms as replacements for the existing standalone platforms.

CallPilot 1.5 Mini

CallPilot 1.5 Mini is a less expensive, embedded messaging stand-alone platform that provides Unified Messaging and base Voice Messaging capabilities in both Meridian 1 and CS 1000S switching environments.

Symposium Call Center Server

Symposium Call Center Server (SCCS) offers a suite of applications that includes call processing and agent handling, extensive management and reporting capabilities, third-party application interfaces, and real-time displays for supervisors and managers. For further information, see the Symposium Call Centre Server documentation.

Nortel Integrated Recorded Announcer

Nortel Integrated Recorded Announcer enables the user to manage recorded announcements using a Browser User Interface (BUI), a Telephone User Interface (TUI), or a text-based user interface. For further information, see *Integrated Recorded Announcer: Service Implementation Guide* (553-3001-360).

Nortel Remote Gateway 9150

Nortel Remote Gateway 9150 enables remote employees of central offices to access CS 1000S features and functionality using an IP WAN. The Remote Gateway 9150 unit installs at the remote site and communicates with the central site using a 10BaseT Ethernet or ISDN Basic Rate Interface (BRI) connection. It uses VoIP technology to route voice and signaling packets between the remote office site and the CS 1000S main site, enabling seamless integration.

The VoIP features automatically switch from the IP network to the PSTN, when voice Quality of Service (QOS) falls below a pre-determined threshold. IP Telephony resumes when QOS levels return to an acceptable level.

Remote Gateway 9150 supports a maximum of 32 digital telephones.

For further information, see *Remote Gateway 9150: Installation and Administration Guide (555-8421-215)*.

Nortel Integrated Call Assistant

The Nortel Integrated Call Assistant can route calls to desired destinations after a series of greetings are played. The destination is based on the caller's telephone keypad strokes. For further information, see *Integrated Call Assistant: Service Implementation Guide (553-3001-362)*.

Nortel Integrated Conference Bridge

The Nortel Integrated Conference Bridge enables a user to schedule and manage conference bridges using a Browser User Interface (BUI) or a Telephone User Interface (TUI). These interfaces enable a user to schedule and configure one-time and recurrent conferences. For further information, see *Integrated Conference Bridge: Service Implementation Guide (553-3001-358)*.

Nortel Integrated Call Director

The Nortel Integrated Call Director enables users to automatically forward incoming telephone calls to another number, such as a cellular or home telephone. The Integrated Call Director continues to forward the call until the call is answered, or all the options are exhausted. For further information, see *Integrated Call Director: Service Implementation Guide (553-3001-361)*.

Nortel Hospitality Integrated Voice Services

Nortel Hospitality Integrated Voice Services provide the hospitality services of Automatic Wake Up (AWU) and Do Not Disturb (DND). The guest dials an access DN and follows instructions until a confirmation message plays to activate a feature. For further information, see *Hospitality Integrated Voice Services: Service Implementation Guide (553-3001-359)*.

Enhanced system management

CS 1000S supports a suite of value-added management capabilities for multiple systems that reduces a customer's total cost of ownership. These include Optivity Telephony Manager (OTM) and Element Manager.

Optivity Telephony Manager

Optivity Telephony Manager (OTM) provides management simplicity and flexible control. OTM features include the following:

- Lightweight Directory Access Protocol (LDAP) directory integration
- station administration
- call accounting
- call tracking
- traffic analysis
- maintenance
- alarm management
- centralized management
- multi-user capability
- web-based functionality
- customizable reporting
- import/export utility
- scheduled tasks
- disaster recovery tools

These features save time and facilities costs, simplifying the management of a complex network.

Element Manager

Element Manager is a simple and user-friendly web-based interface that supports a broad range of system management tasks, including:

- configuration and maintenance of IP Peer and IP telephony features

- configuration and maintenance of traditional routes and trunks
- configuration and maintenance of numbering plans
- configuration of Call Server data blocks (such as configuration data, customer data, Common Equipment data, D-channels)
- maintenance commands, system status inquiries, backup and restore functions
- software download, patch download, patch activation

Element Manager has many features to help administrators manage systems with greater efficiency. Examples are as follows:

- Web pages provide a single point-of-access to parameters that were traditionally available through multiple overlays.
- Parameters are presented in logical groups to increase ease-of-use and speed-of-access.
- The "hide or show information" option enables administrators to see information that relates directly to the task at hand.
- Full-text descriptions of parameters and acronyms help administrators reduce configuration errors.
- Configuration screens offer pre-selected defaults, drop-down lists, checkboxes, and range values to simplify response selection.

The Element Manager web server resides on the Signaling Server and can be accessed directly through a web browser or Optivity Telephony Manager (OTM). The OTM navigator includes integrated links to each network system and their respective instances of Element Manager.

Terminal equipment

The CS 1000S supports the following terminal equipment:

- IP Phones
- 802.11 Wireless LAN
- wireless telephones
- digital telephones

- IP Phone adapter
- analog telephones and devices
- attendant consoles

For further information on analog and digital telephones, see *Telephones and Consoles: Description, Installation, and Operation* (553-3001-367). See *IP Phones: Description, Installation, and Operation* (553-3001-368) for information on IP Phones.

IP Phones

IP Phones provide the desktop clients for CS 1000S IP Telephony. The functionality and call features of these IP Phones are similar to that of a standard digital telephone, such as the M2616. The CS 1000S system supports the following IP Phones:

- Nortel IP Phone 2001
- Nortel IP Phone 2002
- Nortel IP Phone 2004
- Nortel IP Phone 2007
- Nortel IP Audio Conference Phone 2033
- Nortel IP Softphone 2050
- Nortel Mobile Voice Client (MVC) 2050
- WLAN Handsets 2210/2211

802.11 Wireless LAN

CS 1000S supports the 802.11 Wireless LAN. The 802.11 Wireless LAN requires an H.323 Symbol Wireless client terminal.

Wireless telephones

CS 1000S supports 802.11 Wireless IP Gateway, the WLAN IP Telephony Manager 2245 for the WLAN Handsets 2210/2211, and Nortel Integrated DECT (DECT) systems.

802.11 Wireless IP Gateway

The e-mobility Meridian Gateway supports communication between the circuit-switched telephony network and the H.323 Wireless IP terminals on a customer's corporate IP network. For further information, see *WLAN IP Telephony: Installation and Configuration* (553-3001-304).

DECT

The DECT system is an application that enables users to move freely about their work sites using wireless handsets, and keep in communication with end-users and suppliers. Missed calls are reduced and employee productivity is increased due to quick employee response times and multi-tasking flexibility. For further information, see *DECT: Description, Planning, Installation, and Operation* (553-3001-370).

Digital telephones

The CS 1000S system supports a wide range of Digital Line cards that provide global telecommunications market interfaces.

The CS 1000S system supports the M3900 series Meridian Digital Telephones. This includes the M3901 Entry Level Telephone, the M3902 Basic Telephone, the M3903 Enhanced Telephone, the M3904 Professional Telephone, and the M3905 Call Center Telephone.

In conformance with the Evergreen policy, the CS 1000S system also supports other digital telephones, including the M2006, M2008, M2008HF, M2616, M2016S, M2216ACD, and M2317 digital telephones.

IP Phone adapter

The CS 1000S system supports the IP Phone adapter package, for the M26xx and M39xx telephones. The IP Phone adapter package is intended for local deployment and does not support analog PSTN fallback.

Analog telephones and devices

The CS 1000S system supports a wide range of Analog Line cards that provide analog (500/2500-type) telephones and T.38 Fax interfaces.

Attendant consoles

The CS 1000S system supports an Attendant PC software console and the M2250 attendant console.

The Attendant PC software enables users to perform attendant console and call processing functions on a Windows® PC using a mouse or keyboard.

The Attendant PC combines the call-processing power of the M2250 attendant console with the information-processing and storage power of a PC, enhancing attendant services.

Interworking/Interoperability

The CS 1000S uses the standard H.323 protocol to support interworking with other vendors. Interoperability testing is required to determine which products are compatible with CS 1000S systems. A number of products are supported as part of the development, including the following:

- BCM 2.5+
- CS 1000M using an IP Trunk 2.x up-issue

Interworking with Nortel's CS 1000M and BCM products is provided to enable enterprise networks to smoothly migrate from traditional TDM-based PBXs to IP-based telephony products.

System architecture

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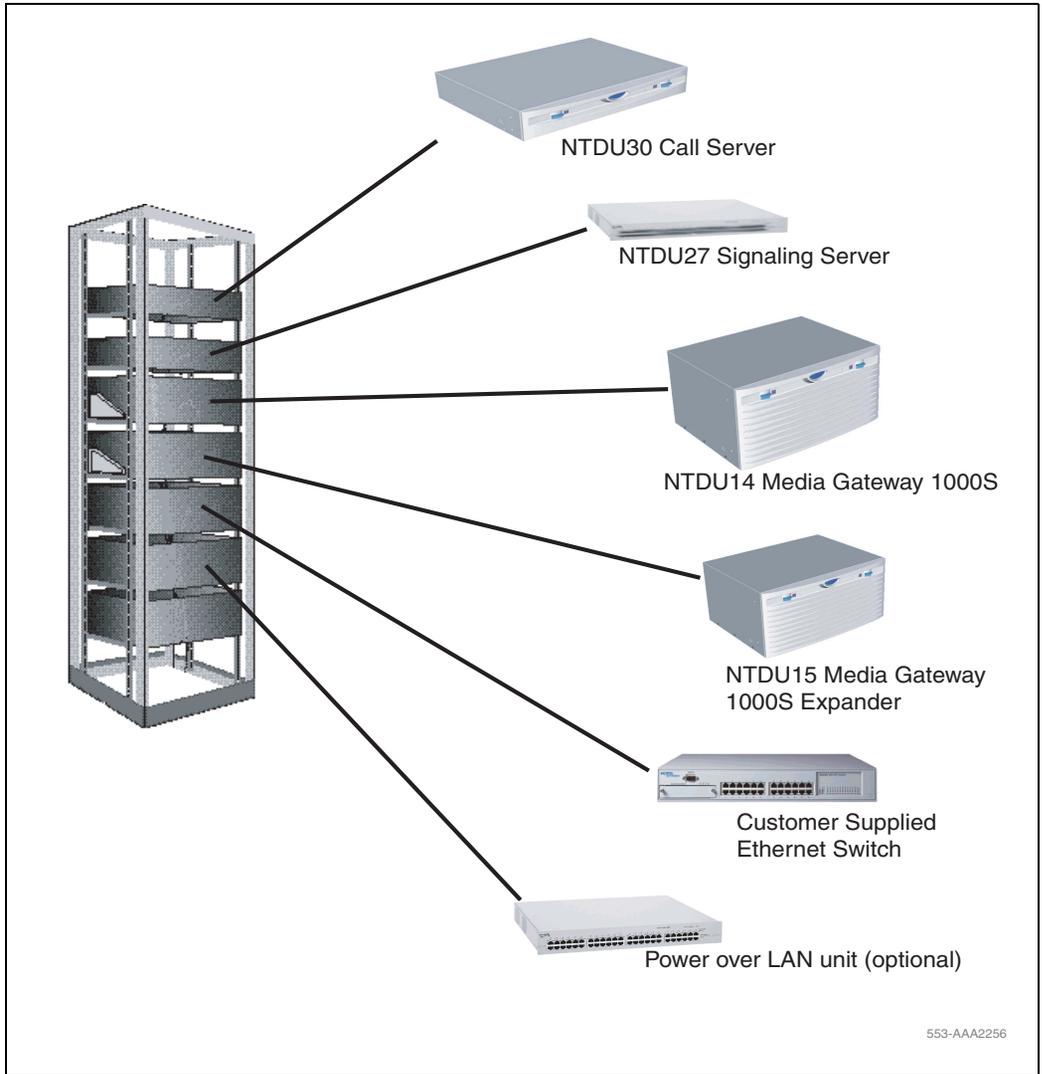
Hardware architecture

A typical CS 1000S system consists of the following major hardware components:

- NTDU30 Call Server
- NTDU27 Signaling Server
- NTDU14 Media Gateway 1000S (MG 1000S)
- NTDU15 Media Gateway 1000S Expander (MG 1000S Expander)
- IP Phones
- customer-supplied Ethernet Layer 2 switch
- Power over LAN unit (optional)

Figure 2 on [page 25](#) shows the major hardware components installed in a customer-supplied 19-inch rack.

Figure 2
CS 1000S components in a rack



Call Server

The Call Server provides telephony services, such as call processing, and supports trunking features. It also acts as a database server for synchronization of configuration information with all MG 1000S systems.

A CS 1000S system can control a maximum of four MG 1000S systems and Expanders.

A CS 1000S system with one Call Server supports up to 1000 IP Phones. You can network a CS 1000S system with other CS 1000S systems to support larger numbers of IP Phones.

Small System Controller card

The NTDK20 Small System Controller (SSC) card in the Call Server is the primary system processor. It controls the telephony services and call processing. Each MG 1000S is also equipped with an SSC card.

The Call Server SSC controls the call-processing features of IP Phones and trunk interfaces when in normal mode of operation.

The Call Server SSC synchronizes the configuration information on all MG 1000S SSC cards. The configuration data on all MG 1000S SSC cards always exactly matches the Call Server SSC configuration data.

The card's Security Device enables the activation of features assigned to the system, through the use of a series of keycodes.

Media Gateway 1000S SSC card

The SSC card in the MG 1000S controls the MG 1000S interface and application cards, and acts as a survivable call processor. All MG 1000S SSC cards are synchronized with the call processing on the Call Server SSC card. This synchronization enables an MG 1000S SSC card to take over local call processing if the primary Call Server fails to respond.

If the Call Server fails to respond, each MG 1000S SSC can become its own independent call processor. This means that in the event of Call Server failure, one MG 1000S SSC does not act as a Call Server for the rest of the system; they are all independent.

Signaling Server

CS 1000S systems use a Signaling Server. The Signaling Server is an industry-standard, PC-based server that provides a central processor to drive H.323 and Session Initiation Protocol (SIP) signaling for IP Phones and IP Peer Networking. It provides signaling interfaces to the IP network using software components that operate on the VxWorks™ real-time operating system.

The Signaling Server has both an ELAN and TLAN network interface. The Signaling Server communicates with the Call Server through the ELAN subnet.

The Signaling Server is mounted in a 19-inch rack. The Signaling Server can be installed in a load-sharing redundant configuration for higher scalability and reliability.

The following software components operate on the Signaling Server:

- Terminal Proxy Server
- SIP/H.323 Signaling Gateways
- Network Routing Service
- Element Manager

All the software elements can coexist on one Signaling Server or reside individually on separate Signaling Servers, depending on traffic and redundancy requirements for each element.

For more information about H.323 and SIP Trunking, refer to *IP Peer Networking: Installation and Configuration* (553-3001-213).

Terminal Proxy Server

The Terminal Proxy Server (TPS) acts as a signaling gateway between the IP Phones and the Call Server using the UNISim protocol. It performs the following functions:

- converts the IP Phone UNISim messages into messages the Call Server can interpret
- allows IP Phones to access telephony features provided by the Call Server

The TPS also controls the IP Phone registration.

SIP/H.323 Signaling Gateways

SIP/H.323 Signaling Gateways are software components configured on virtual loops, similar to IP Phones. SIP/H.323 Signaling Gateways bridge existing call processing features and the IP network. They also enable access to the routing and features in the MCDN feature set.

Note 1: The SIP/H.323 Signaling Gateway must register with the Network Routing Service (NRS).

Note 2: Virtual TNs enable you to configure service data without hardwiring IP Phones to the CS 1000S system. Virtual TNs are configured in LD 97.

To support IP Peer Networking, dual Call Servers in a CS 1000S must be associated with Signaling Servers that run SIP/H.323 Signaling Gateway software. The number of Signaling Servers required depends on the capacity and level of redundancy required.

Network Routing Service

The NRS application provides network-based routing, combining the following into a single application:

- **H.323 Gatekeeper** — The H.323 Gatekeeper provides central dialing plan management and routing for H.323-based endpoints and gateways.
- **SIP Redirect Server** — The SIP Redirect Server provides central dialing plan management and routing for SIP-based endpoints and gateways.

- **NRS Database** — The NRS database stores the central dialing plan in XML format for both the SIP Redirect Server and the H.323 Gatekeeper. The SIP Redirect Server and H.323 Gatekeeper both access this common endpoint and gateway database.
- **Network Connect Server (NCS)** — The NCS is used only for Media Gateway 1000B (MG 1000B) and Virtual Office solutions.
- **NRS Manager web interface** — The NRS provides its own web interface to configure the SIP Redirect Server, the H.323 Gatekeeper, and the NCS.

The NRS application provides routing services to both H.323- and SIP-compliant devices. The H.323 Gatekeeper can be configured to support H.323 routing services, while the SIP Redirect Server can be configured to support SIP routing services. The H.323 Gatekeeper and the SIP Redirect Server can reside on the same Signaling Server.

Each system in an IP Peer network must register to the NRS. The NRS software identifies the IP addresses of systems based on the network-wide numbering plan. NRS registration eliminates the need for manual configuration of IP addresses and numbering plan information at every site.

Element Manager

Element Manager is a software application that provides a web interface to configure and maintain system components, including the Signaling Server. Single web pages provide access to information traditionally spread into multiple overlays.

For more information on the Element Manager application, see “Element Manager” on [page 18](#).

Media Gateway 1000S

The MG 1000S holds Intelligent Peripheral Equipment (IPE) cards and application cards. In a normal operational state, the Call Server controls the MG 1000S. The MG 1000S processor provides low-level control of the interface cards installed in the MG 1000S slots, and communicates with the Call Server for feature operation.

The MG 1000S supports a Small System Controller (SSC) card and four slots for flexible configurations of line, trunk, and application cards.

Note: You can configure the SSC in the MG 1000S for survivability to assume local call processing if the Call Server cannot be accessed. If the MG 1000S is not configured as survivable, then the MG 1000S is out-of-service until the Call Server is again accessible.

Voice Gateway Media Cards are installed in the MG 1000S. A Voice Gateway Media Card provides transcoding between IP and circuit-switched protocol.

Each MG 1000S can support one MG 1000S Expander.

Ethernet switch (customer-supplied)

The customer-supplied Layer 2 Ethernet switch transmits data packets to interconnected Ethernet-attached devices. The switch directs the data only to the targeted device, rather than to all attached devices. See *Data Networking for Voice over IP* (553-3001-160) for details.

Power over LAN (optional)

An optional Power over LAN unit adds power and data communication over standard Category 5 LAN drops for powering IP Phones. The Power over LAN unit eliminates the need to connect each telephone to an AC power outlet. This saves in desktop wiring and enables the use of a centralized Uninterruptible Power Supply (UPS) for power backups.

Infrastructure

Various infrastructure components are required to support VoIP. These components include switches, routers, and MG 1000S. The data network infrastructure's engineering and provisioning is critical to achieve satisfactory telephony voice quality. For more information, see *Data Networking for Voice over IP* (553-3001-160).

Software architecture

Centralized Automatic Software Upgrade

Centralized Automatic Software Upgrade allows a new version of software to be automatically loaded on the MG 1000S after the Call Server software is upgraded. To reduce service impact, a sequential upgrade mode upgrades only one MG 1000S at a time.

Centralized upgrade

Centralized upgrade enables the IP telephony components (Voice Gateway Media Cards) to be upgraded from the Element Manager interface. The Element Manager host, the Signaling Server, is a file server for software upgrade files for the Voice Gateway Media Cards. IP Phones also automatically upgrade from a centralized file location (their TPS).

Centralized patching

Centralized patching enables IP telephony components, the Call Server, and MG 1000S systems, to be patched from a central location through Element Manager.

File uploading

File uploading enables software upgrade files and patches to be uploaded to the Element Manager host, the Signaling Server, for centralized upgrading or centralized patching. The file is uploaded from the management PC (web browser) to the Element Manager host (web server).

Patching implementation

The Call Server, MG 1000S, Signaling Server, and the Voice Gateway Media Cards are patchable. Installing a patch enables a fix to be delivered, without requiring a new version of software.

Configuration options

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Introduction

The IP-distributed architecture of CS 1000S offers the benefits of location flexibility for system components. The architecture also provides wiring control and simplification of space planning, system management, and databases.

CS 1000S bridges the circuit-switched PBX and the IP network, or it can operate in an IP-only network. This consolidation of network traffic simplifies the task of managing communication to a single, efficient IP network. End users can deploy the CS 1000S in many flexible configurations

in a LAN or WAN environment. Although different combinations are possible, most installations fall into one of the following categories:

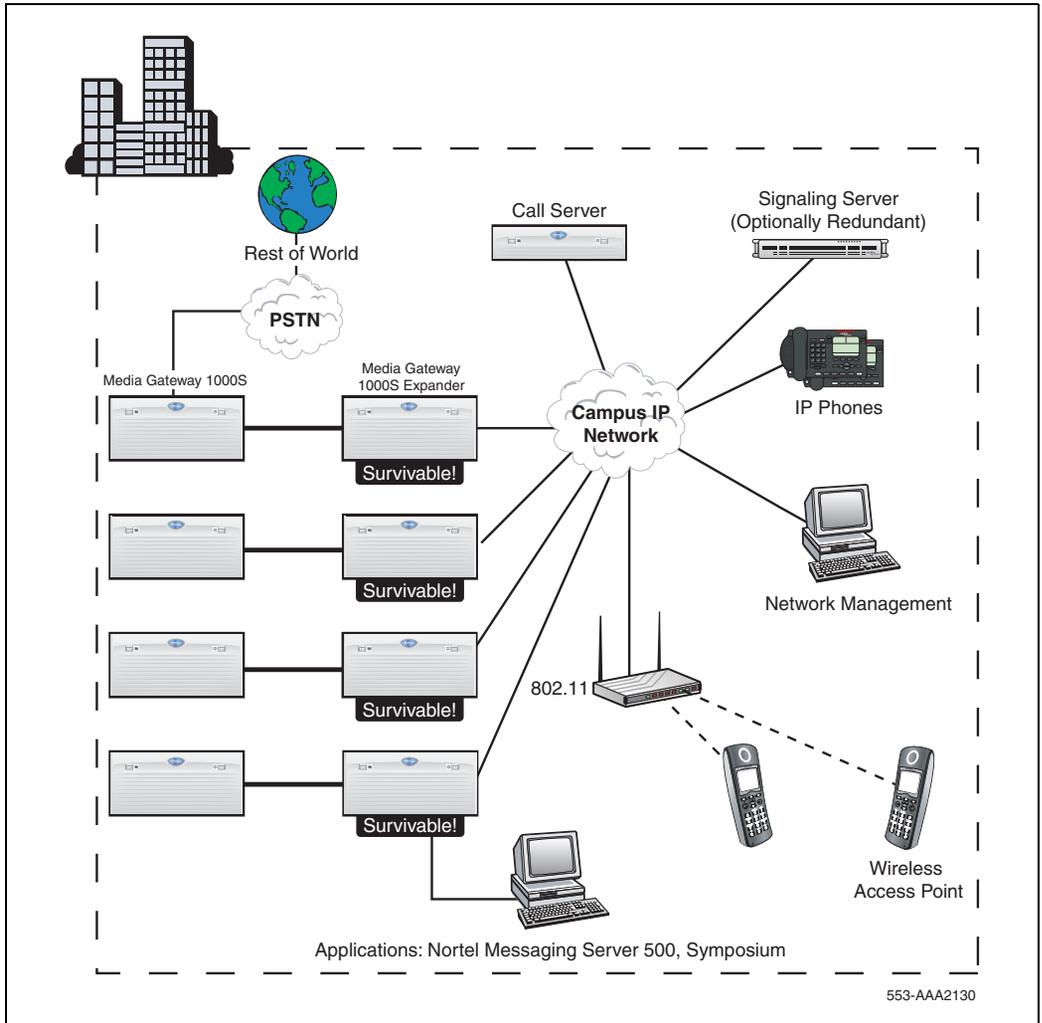
- Single Site
 - single building
 - multiple buildings in a campus
- Multiple Sites
 - central Call Server with branch office users
 - distributed Call Servers

For more information on system deployment options for the CS 1000S, refer to *Data Networking for Voice over IP* (553-3001-160).

Single campus deployment

Figure 3 shows an example of a single campus deployment and its components.

Figure 3
Single campus deployments example



Note: You can configure the network for redundancy.

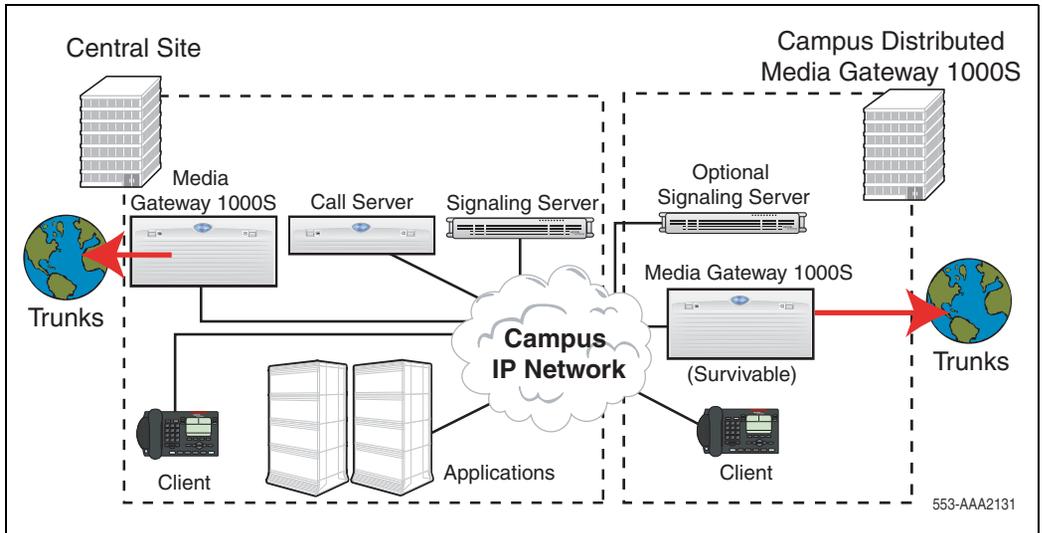
The single campus deployment example consists of the following:

- Call Server
- Signaling Server (optionally redundant)
- up to four Media Gateways and four Media Gateway Expanders:
 - can be configured to provide network redundancy
 - provide a global suite of telephony trunks
- analog telephones (service for 2500 series telephones and fax)
- digital telephones (service for Meridian 3900 telephones)
- a switched Ethernet infrastructure
- up to 1000 IP Phones
- centralized management capability
- other telephony applications

Multiple buildings in a campus

Figure 4 shows a CS 1000S system distributed across multiple buildings in a campus setting. You can distribute MG 1000S systems and users across a campus IP Network.

Figure 4
Multi-site deployment with centralized call processing



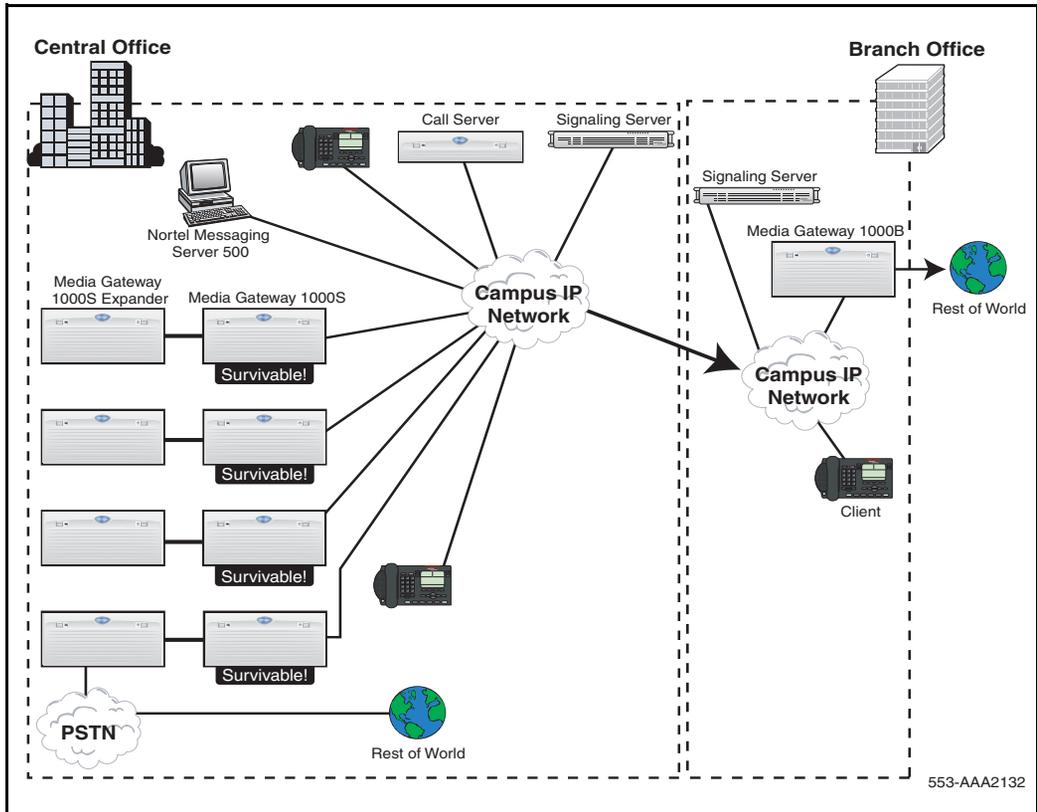
In Figure 4, the multi-site deployment with centralized call processing consists of the following:

- Call Server at the central site
- Campus distributed MG 1000S
 - optional Signaling Server
 - MG 1000S (survivable)
 - IP Phones configured and managed centrally from the central site
 - Local trunks PSTN and 911 access.

Multiple sites

Figure 5 shows a multi-site deployment with a central Call Server and a branch office.

Figure 5
Central Call Server with branch office



The following apply to a multi-site deployment:

- The Call Server at the central site that provides call processing for the main and branch offices.
- The multi-site deployment can contain a branch office.

- Requires its own Signaling Server.
- MG 1000B in remote branch offices enables local PSTN and 911 access.
- IP Phones are configured and managed centrally from the main office.
- The MG 1000B is survivable in the event of a WAN failure.
- You can configure up to 400 users for survivability.
- You can use the Signaling Server virtual trunking to the central site for system management purposes.

Routing to and from local branch offices

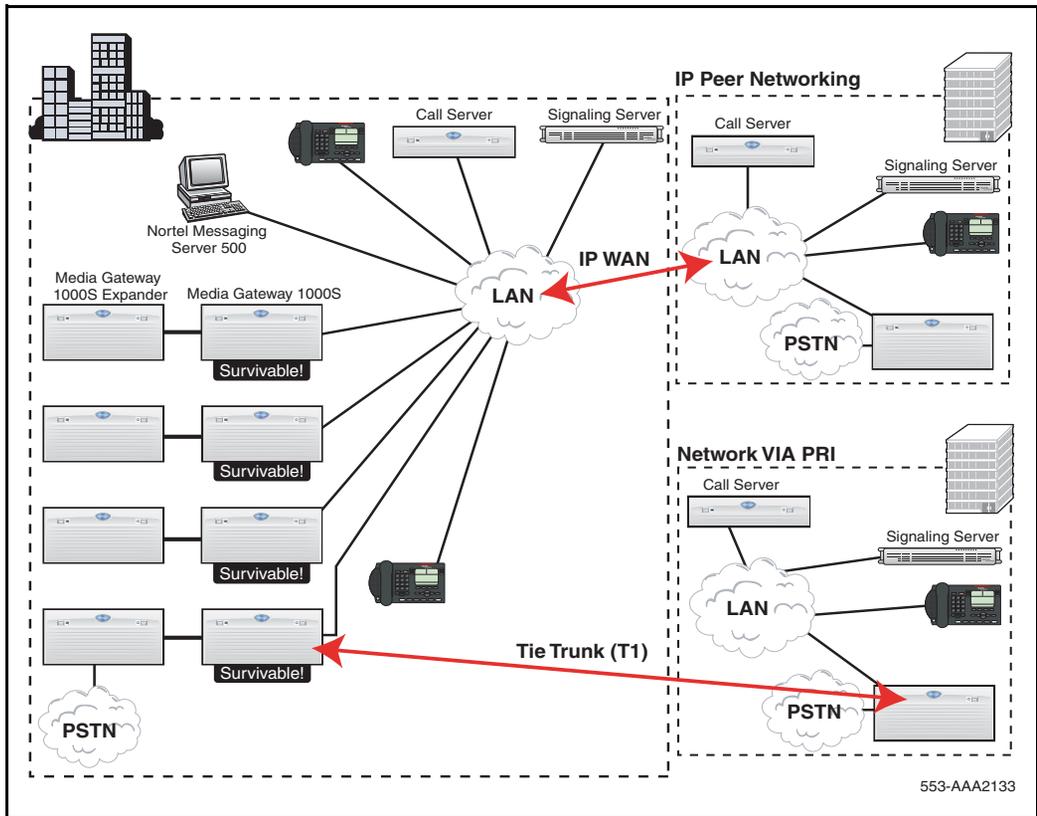
IP Phone users can be located at a branch office that is equipped with an MG 1000B Core. Call routing to the local MG 1000B Core is important, especially if toll charges are applicable to calls from the central Call Server controlling the IP Phone. The administrator can configure digit manipulation for IP Phones that are located near a branch office, in order to select a branch office that provides local PSTN access.

Calls from the PSTN to users within the network can be routed using either (a) the various ESN numbering plan configurations, or (b) the Network Number Resolution feature, based upon the Vacant Number Routing feature. This enables small sites, such as a branch office, to require minimal configuration to route calls through other Call Servers or the Gatekeeper. See *Branch Office: Installation and Configuration* (553-3001-214).

Distributed Call Servers

Distributed Call Servers are defined as multiple CS 1000S systems. Figure 6 shows an example of multi-site deployment with distributed Call Servers.

Figure 6
Distributed Call Servers



The multi-site deployment consists of the following peer sites:

- separate Call Servers that provide call processing for each site
- networking supported with IP data network or circuit-switched (analog or digital) trunks

- full-featured networking between sites using IP Peer Networking and/or TIE trunks
- transparent to user at all sites

Reliability strategies

Communication reliability is critical to the operation of any business. A number of capabilities are available in CS 1000S to ensure that telephony is available when one or more of the following situations occur:

- a hardware component fails
- a software component fails
- the IP network suffers an outage

The CS 1000S provides several levels of redundancy to ensure that the telephony services can withstand single hardware and network failures. CS 1000S provides the following component redundancy:

- Call Server with automatic database distribution
- Signaling Server software, including H.323 Gateway and IP Phone software
- H.323 Gatekeeper
- H.323 Gateway interface to Gatekeeper
- Campus-distributed MG 1000S in Survival Mode

You can provision a CS 1000S to provide the following component redundancy:

- If the Call Server is unavailable, you can configure MG 1000S systems as survivable independent systems (automatic database synchronization).

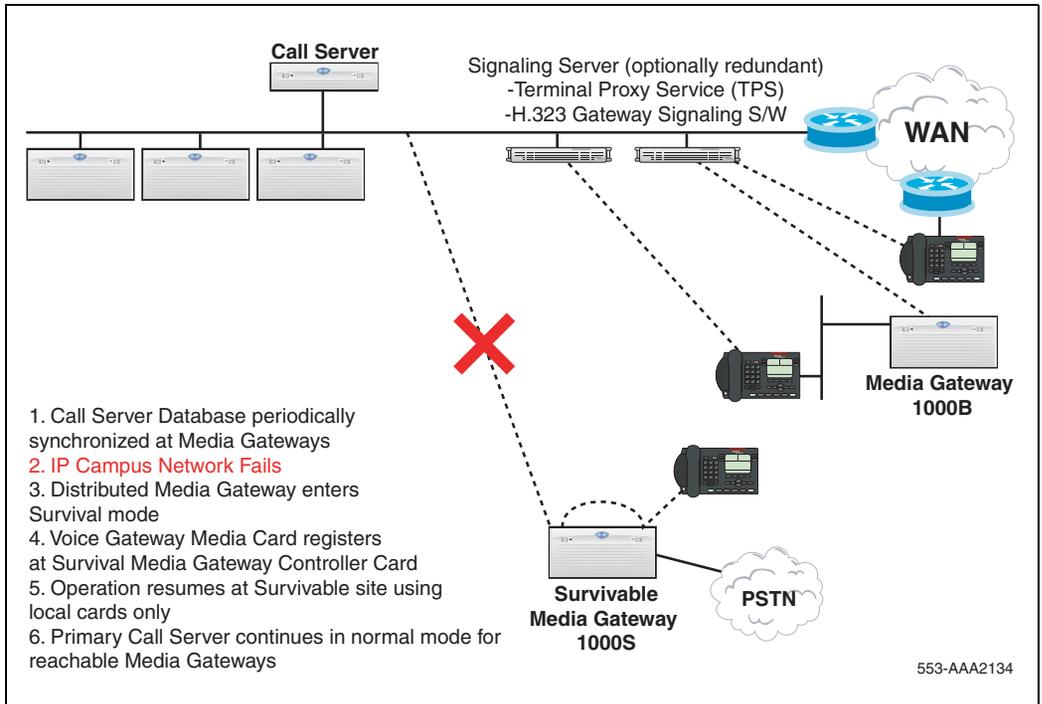
Note: If the MG 1000S is not configured as survivable, then it is out of service until the Call Server is in service again.

- If the Call Server is unavailable, the IP telephony node can re-register with one MG 1000S specified as an Alternate Call Server.

- If the master (Leader) Signaling Server is unavailable, a Follower Signaling Server becomes the IP Telephony node master.
- Signaling Servers share telephone registrations (load-sharing).
- If a virtual trunk is unavailable, virtual trunk redundancy occurs (that is, the call-processing software selects an alternate route).
- Signaling Servers can be configured for different types of redundancy for their software components (for example, Alternate Gatekeeper, Failsafe Gatekeeper).
- If the Primary Gatekeeper is unavailable, you can configure an Alternate Gatekeeper (automatic database synchronization).
- If the Primary and Alternate Gatekeeper are unavailable, you can configure a Failsafe Gatekeeper (at each Signaling Server).

Figure 7 on [page 43](#) illustrates the handling of a network failure with Survivable MG 1000S in place.

Figure 7
Network failure with survivable MG 1000S systems



IP telephony node configuration

An IP telephony node is a grouping of Voice Gateway Media Cards (and Signaling Servers), regardless of the location of the Voice Gateway Media Cards in the MG 1000S. Therefore, several MG 1000S can belong to the same node. Alternately, each MG 1000S can have its own node.

Each IP telephony node can be configured with the IP address of an Alternate Call Server, to which it registers if the Call Server is unavailable.

Note: The Alternate Call Server is an MG 1000S SSC that is configured as survivable.

The survivable MG 1000S (Alternate Call Server) IP address must be on the same ELAN subnet as the Call Server. If the MG 1000S is on a physically different subnet, such as in a different building, then you can use VLANs to keep IP addresses on the same logical subnet. For further implementation details, refer to *Data Networking for Voice over IP* (553-3001-160).

If there are different nodes in different MG 1000S, then the nodes can be configured to register to different Alternate Call Servers. This concept is desirable for optimizing system reliability to best deal with possible system outages. Associate each IP telephony node with an appropriate (for example, co-located) Alternate Call Server.

If the node IDs are configured using the guidelines for the 'Enhanced Redundancy for IP Line Nodes' feature, then the IP Phones can register (if needed) to an alternate node on an MG 1000S Expander. This further improves the survivability of the IP Phones by allowing them to register to a different node should a system outage occur on their primary node's MG 1000S.

Refer to *IP Line: Description, Installation, and Operation* (553-3001-365) for a description of the Enhanced Redundancy for IP Line Nodes feature.

Nortel Communication Server 1000
Communication Server 1000S
Overview

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