Nortel Communication Server 1000

Communication Server 1000E Planning and Engineering

NN43041-220

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How to get help

This section explains how to get help for Nortel products and services.

Finding the latest updates on the Nortel Web site

The content of this documentation was current at the time the product was released. To check for updates to the latest documentation and software for Nortel Communication Server 1000E (CS 1000E), click one of the following links.

Latest Software	Takes you directly to the Nortel page for CS 1000E software.
Latest Documentation	Takes you directly to the Nortel page for CS 1000E documentation.

Getting help from the Nortel Web site

The best way to get technical support for Nortel products is from the Nortel Technical Support Web site:
www.nortel.com/support

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. More specifically, the site enables you to:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

Getting help over the telephone from a Nortel Solutions Center

If you do not find the information you require on the Nortel Technical Support Web site, and have a Nortel support contract, you can also get help over the phone form a Nortel Solutions Center.

In North America, call 1-800-4NORTEL (1-800-466-7835).

Outside North America, go to the following Web site to obtain the phone number for your region:
www.nortel.com/callus

Getting help from a specialist by using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to:
www.nortel.com/erc

Getting help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or authorized reseller, contact the technical support staff for that distributor or reseller.

New in this release

The following sections detail what’s new in *Communication Server 1000E Planning and Engineering (NN43041-220)* for Nortel Communication Server 1000 (CS 1000) Release 5.0.

Contents

This section contains information about the following topics:

"Media Gateway Controller" (page 13)

"CP PM Call Server" (page 14)

"CP PM Signaling Server" (page 15)

"COTS Signaling Server" (page 15)

"Media Card 32S" (page 16)

"Geographic Redundancy Survivable Media Gateway" (page 16)

"Network Music" (page 17)

"About this document" (page 17)

Media Gateway Controller

The Media Gateway Controller (MGC) card occupies the system controller slot 0 in the Media Gateway chassis. MGC is an upgrade over the NTDK20 Small System Controller card (SSC).

The NTDW60 MGC card provides a gateway controller for IP Media Gateways in a CS 1000E system. The MGC only functions as a gateway controller under control of a CS 1000E (CP PIV or CP PM) call server.

The MGC provides the following key features:

- Increased processing power (10x over SSC)
- Increased memory capacity (128 MB vs 32 MB on SSC)

- Compact Flash used for permanent storage
- 2 expansion daughterboard sites
- Embedded Layer 2 switch supports enhanced dual-homing
- Enables co-resident applications (Voice Gateway and Media Gateway Controller)
- Enhanced diagnostics
- Enhanced loadware patching
- Increased reliability

The MGC features allow for a reduction of CS 1000E hardware:

- Replacement of the SSC card in the MG 1000E. SSC is still supported in a CP PII or CP PIV based CS 1000E.
- Reduction of Voice Gateway Media Cards with the use of MGC Digital Signal Processor daughterboards.
- Replacement of MG 1000T peer gateways. MG 1000E with MGC supports PRI/PRI2/DTI/DTI2 trunks, BRI trunks, D-channels, and clock controllers.
- Reduction of a separate Terminal Servers with the MGC remote SDI feature.

Digital Signal Processor daughterboard

The MGC card supports up to 128 DSP channels (equivalent of 4 Media Cards) with the two expansion sites to accommodate digital signal processor daughterboards. 96-port DSP daughterboard NTDW64 and 32-port DSP daugherboard NTDW62 can be installed on the MGC. The DSP daughterboards support Voice over IP (VoIP) voice gateway resources on the MGC, reducing the need for separate Voice Gateway Media Cards.

The MGC DSP daughterboard security feature provides an infrastructure to allow endpoints capable of SRTP/SRTCP to engage in secure media exchanges. The media security feature can be configured by the administrator or, optionally by the end user. This feature provides for the exchange of cryptographic material needed by the SRTP-capable endpoints to secure media streams originating from those endpoints.

For more information about MGC and DSP daughterboards, see "Media Gateway Controller (MGC) card" (page 54).

CP PM Call Server

The CP PM is a new hardware platform on an IPE slot form factor, allowing for a CS 1000E product with only a single MG 1000E chassis or cabinet.

The CP PM is a Pentium Mobile based server that provides features similar to the CP PIV/CP PII processor. One major difference is that unlike the CP PIV/CP PII, the CP PM Call Server solution does not have an external System Utility (SUTL) card. The CP PM provides similar SUTL functionality without requiring any external hardware.

CP PM Call Server only supports the Media Gateway Controller (MGC) as the IPMG controller. Small system main and branch office systems that are migrated to CP PM Call Server require that all the SSC cards be replaced with MGC cards.

For more information about CP PM Call Server, see "CP PM Call Server hardware components" (page 51).

CP PM Signaling Server

The CP PM Signaling Server feature takes the existing ISP1100 Signaling Server software and ports it to the new CP PM hardware platform.

CP PM Signaling Servers provide the following functionality and enhancements:

- Signaling Server applications run on VxWorks on the new CP PM hardware.
- New installation to make use of the Fixed Media Device (FMD) software pre-loaded on FMD.
- Software updates for CP PM Signaling Server are available with the use of the new Removable Media Device (RMD) Compact Flash. The RMD can also be used for backup and restore of IP configuration.
- The inventory command enhancement on the Call Server is modified to include the CP PM Signaling Server.
- Faceplate LED functionality with enhanced visual indicators.
- Patching is enhanced so Signaling Server applications software patches apply to CP PM Signaling Server.
- The partition size of the FMD for the CP PM Signaling Server is increased compared to the ISP 1100.

For more information about CP PM Signaling Server, see "CP PM Signaling Server hardware components" (page 73).

COTS Signaling Server

The Signaling Server on COTS servers program takes the existing ISP1100 Signaling Server software and ports it to two new platforms: the HP-DL320-G4 server and the IBM-X306m server.

With this feature:

- Signaling Server applications run on VxWorks on the COTS Signaling Server.
- New installation to make use of the Fixed Media Device (FMD) software pre-loaded on FMD.
- Software updates for COTS Signaling Server are available with the use of Universal Serial Bus (USB) flash drives. COTS Signaling Server USB ports can also be used for backup and restore of IP configuration.
- The inventory command enhancement on Call Server is modified to include the COTS Signaling Servers.
- Patching is enhanced so Signaling Server application software patches apply to COTS Signaling Server hardware.
- The partition size of the FMD for the COTS Signaling Server is increased compared to the ISP1100.

For more information on COTS Signaling Servers, see "COTS Signaling Server hardware components" (page 75)

Media Card 32S

The Media Card 32S (MC32S) card is added to the CS 1000 portfolio. This card is an enhancement over the existing Media Card 32 to allow for SRTP (Secure Real Time Protocol). SRTP is used to secure the IP media path to and from the DSP channels on the MC32S.

The new MC32S card provides 32 channels of IP-TDM connectivity between an IP device and a TDM device in the CS 1000 network. The MC32S supports the same applications as the MC32. IP trunk is not supported.

The MC32S card comes in an IPE form factor and can run in any IPE slot on a CS 1000 Release 5.0 system. The MC32S can interwork with other voice gateway application cards such as the MGC, ITG-SA, and ITG-Pentium cards.

For more information about the Media Card 32S, see Figure 19 "Media Card 32S" (page 67).

Geographic Redundancy Survivable Media Gateway

Geographic Redundancy Survivable Media Gateway is an enhancement of the current Geographic Redundancy feature, which allows for the new capabilities defined below:

- Provides support for up to 50 secondary Call Servers which can be spread around the customer network for redundancy.

- Provides Media Gateway Time Division Multiplexing (TDM) resource survivability by allowing triple registration of MGC and Media Cards to the primary, alternate 1, and alternate 2 Call Servers.
- Provides the ability to install Media Gateways geographically dispersed over a customers network.

For more information about Geographic Redundancy Survivable Media Gateway, see "Option 4: Geographic Redundancy Survivable Media Gateway" (page 85).

Network Music

Network Music is an enhancement over the existing music service. Existing music service was provided from a local source. This meant every node required its own music source.

With the Network Music service, the CS 1000 system support Music On Hold without a locally equipped music source. Network Music accesses a central audio server in the network through H.323/SIP virtual trunks or TDM trunks.

To maximize resource efficiency, the music is broadcast so that multiple parties can share the same music trunk. One music trunk can support a maximum of 64 listeners with broadcast music.

For more information about Network Music, see "Network Music" (page 198).

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 1000E 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, see *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

This document provides the information necessary to properly engineer a Communication Server 1000E (CS 1000E) system. There are two major purposes for using this document: to engineer an entirely new system, and to evaluate a system upgrade.

The EC provides an alternative to the manual processes given in this document. It is beyond the scope of this document to describe the EC process.

Note on legacy products and releases

This NTP contains information about systems, components, and features that are compatible with Nortel Communication Server 1000 Release 5.0 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 1000E (CS 1000E) system.

When upgrading software, memory upgrades can be required on the Signaling Server, the Call Server, or both.

Intended audience

This document is intended for system engineers responsible for engineering the switch and the Nortel Technical Assistance Support personnel who support them. Engineers can be employees of the end user, third-party consultants, or distributors.

The engineer responsible for system implementation should have several years of experience with Nortel PBX systems.

Others who are interested in this information, or find it useful, are Sales and Marketing, Service Managers, Account Managers, and Field Support.

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Feature Listing Reference (NN43001-111)*
- *Converging the Data Network with VoIP Fundamentals (NN43001-260)*

- *Electronic Switched Network Signaling and Transmission Guidelines (NN43001-280)*
- *Transmission Parameters (NN43001-282)*
- *Dialing Plans Reference (NN43001-283)*
- *Circuit Card Reference (NN43001-311)*
- *Signaling Server Installation and Commissioning (NN43001-312)*
- *IP Peer Networking Installation and Commissioning (NN43001-313)*
- *Branch Office Installation and Commissioning (NN43001-314)*
- *Automatic Call Distribution Fundamentals (NN43001-551)*
- *Network Routing Service Installation and Commissioning (NN43001-564)*
- *System Management Reference (NN43001-600)*
- *Access Control Management Reference (NN43001-602)*
- *Software Input Output Administration (NN43001-611)*
- *Telephony Manager 3.1 System Administration (NN43050-601)*
- *Telephony Manager 3.1 Telemanagement Applications Fundamentals (NN43050-602)*
- *Security Management (NN43001-604)*
- *Element Manager System Reference - Administration (NN43001-632)*
- *IP Line Fundamentals (NN43100-500)*
- *Telephones and Consoles Fundamentals (NN43001-567)*
- *IP Phones Fundamentals (NN43001-368)*
- *ISDN Primary Rate Interface Fundamentals (NN43001-569)*
- *Basic Network Feature Fundamentals (NN43001-579)*
- *ISDN Basic Rate Interface Feature Fundamentals (NN43001-580)*
- *Traffic Measurement Formats and Outputs Reference (NN43001-750)*
- *Software Input Output Reference - Maintenance (NN43001-711)*
- *Communication Server 1000M and Meridian 1 Large System Planning and Engineering (NN43021-220)*
- *Communication Server 1000E Installation and Commissioning (NN43041-310)*
- *Communication Server 1000E Upgrades (NN43041-458)*
- *CallPilot Planning and Engineering (555-7101-101)*

Online

To access Nortel documentation online, click the **Technical Documentation** link under **Support & Training** on the Nortel home page:

www.nortel.com

CD-ROM

To obtain Nortel documentation on CD-ROM, contact your Nortel customer representative.

Overview of the engineering process

Contents

This section contains information on the following topics:

"Introduction" (page 21)

"Engineering a new system" (page 22)

"Engineering a system upgrade" (page 22)

"EC" (page 24)

Introduction

WARNING
Before a CS 1000E 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, see *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

A switch must be engineered upon initial installation, during upgrades, and when traffic loads change significantly or increase beyond the bounds anticipated when the switch was last engineered. A properly engineered switch is one that all components work within their capacity limits during the busy hour.

This document is not intended to provide a theoretical background for engineering principles, except to the extent required to make sense of the information. Furthermore, in order to control complexity, technical details and data are sometimes omitted when the impact is sufficiently small.

This document does not address the engineering or functionality of major features, such as Automatic Call Distribution (ACD) or Network Automatic Call Distribution (NACD), and of auxiliary processors and their applications, such as Symposium and CallPilot. Guidelines for feature and auxiliary platform engineering are given in documents relating to the specific applications involved. This document provides sufficient information to determine and account for the impact of such features and applications upon the capacities of the system itself.

Engineering a new system

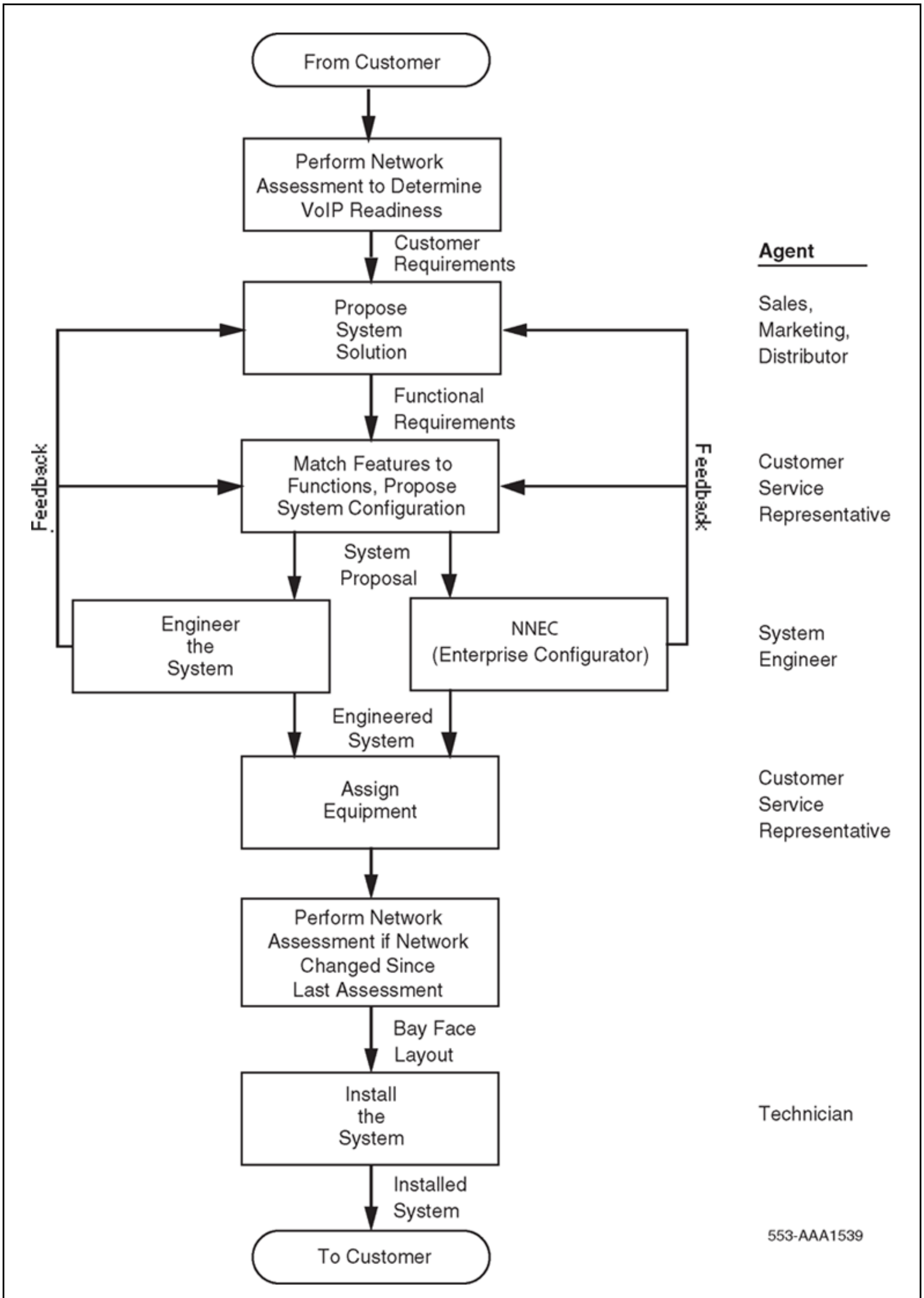
Figure 1 "Engineering a new system" (page 23) illustrates a typical process for installing a new system. The agent expected to perform each step of the process is listed to the right of the block. The highlighted block is the subject of this document.

Engineering a system upgrade

In cases of major upgrades or if current resource usage levels are not known, Nortel recommends following the complete engineering process, as described for engineering a new system.

If minor changes are being made, calculate the incremental capacity impacts and add them to the current resource usage levels. Then compare the resulting values with the system capacities to determine whether the corresponding capacity has been exceeded.

Figure 1
Engineering a new system



EC

The EC is a global engineering and quotation tool to assist the site engineer, sales person, or customer in engineering the switch. It is available in both stand-alone and web-based versions. For users in North America and the Caribbean and Latin America (CALA), it replaces Meridian Configurator and 1-Up. For users in Europe, Middle East, and Africa (EMEA) countries, it replaces NetPrice.

The EC provides a simple "needs-based" provisioning model that allows for easy configuring and quoting. The EC supports CS 1000E new system sales and upgrades by analyzing input specifications for a digital PBX to produce a full range of pricing, engineering reports, and graphics. These reports include equipment lists, cabling reports, software matrix, engineering capacities, and pricing for currently available CS 1000E configurations. Graphics depict the engineered platform, card slot allocations as well as loop assignments.

The EC runs on the user’s Windows-based or MacOS personal computer. It uses standard browser and Microsoft Office applications. For details on computer system requirements and for user instructions, refer to the Nortel web site. EC implements the algorithms specified in this document for real time, memory, and physical capacities. It is the official tool for determining whether a proposed configuration will meet the customer’s capacity requirements.

Where applicable, in this document, references are made to the EC inputs that correspond to parameters being described.

Regulatory information

Contents

This section contains information on the following topics:

"System approval" (page 25)

"Electromagnetic compatibility" (page 26)

"Notice for United States installations" (page 27)

"Notice for Canadian installations" (page 29)

"Canadian and US network connections" (page 30)

"Notice for International installations" (page 32)

"Notice for Germany" (page 32)

System approval

The CS 1000E system has approvals to be sold in many global markets. Regulatory labels on the back of system equipment contain national and international regulatory information.

Some physical components in systems have been marketed under different names in the past. Previous naming conventions utilizing the terms *Succession 1000* and *CSE 1000* have been harmonized to use the term *Communication Server 1000*. Similarly, previous naming conventions utilizing the terms *Meridian* and *Option* have been harmonized to use the term *Meridian 1 PBX*. Product names based on earlier naming conventions can still appear in some system documentation and on the system regulatory labels. From the point of view of regulatory standards compliance, the physical equipment is unchanged. As such, all the instructions and warnings in the regulatory sections of this document apply to the Communication Server 1000M, Communication Server 1000S, and Communication Server 1000E systems, as well as the Meridian, Succession 1000, and CSE 1000 systems.

Electromagnetic compatibility

CAUTION
In a domestic environment, the system can cause radio interference. In this case, the user can be required to take adequate measures.

Table 1 "EMC specifications for Class A devices" (page 26) lists the EMC specifications for the system.

Table 1
EMC specifications for Class A devices

Jurisdiction	Standard	Description
United States	FCC CFR 47 Part 15	FCC Rules for Radio Frequency Devices (see Note 1a)
Canada	ICES-003	Interference-Causing Equipment Standard: Digital Apparatus
Europe	EN 55022/ CISPR 22	Information technology equipment — Radio disturbance characteristics — Limits and methods of measurement (see Note 2)
	EN 55024	Information technology equipment — Immunity characteristics — Limits and methods of measurement
	EN 61000-3-2	Limits for harmonic current emissions (equipment input current <= 16 A per phase)
	EN 61000-3-3	Limitation of voltage fluctuations and flicker in low-voltage supply systems for equipment with rated current <= 16 A
Australia	CISPR 22/ AS/NZS 3548	Limits and methods of measurement of radio disturbance characteristics of information technology equipment (see Note 2)
Korea	KN22	Information technology equipment — Radio disturbance characteristics — Limits and methods of measurement
	KN24	Information technology equipment — Immunity characteristics — Limits and methods of measurement
Taiwan	CNS 13438	Limits and methods of measurement of radio disturbance characteristics of information technology equipment

Jurisdiction	Standard	Description
Note 1a: FCC CFR 47 Part 15.21 statement: "Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, can cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user is required to correct the interference at his own expense." ***Note 1b:*** The user should not make changes or modifications not expressly approved by Nortel. Any such changes can void the user's authority to operate the equipment.		
Note 2: EN 55022/CISPR 22 statement: "WARNING This is a class A product. In a domestic environment this product can cause radio interference, in which case the user can be required to take adequate measures."		

Notice for United States installations

The system complies with Part 68 of the United States Federal Communications Commission (FCC) rules. A label containing the FCC registration number and Ringer Equivalence Number (REN) for the equipment is on the back of each Media Gateway and Media Gateway Expander. If requested, you must provide this information to the telephone company.

Regulatory labels include:

- FCC registration: AB6CAN-61117-MF-E
- FCC registration: AB6CAN-61116-PF-E
- FCC registration: AB6CAN-18924-KF-E
- Service code: 9.0F, 6.0P
- Ringer equivalence (REN): 2.7A

The FCC regulation label includes the REN. This number represents the electrical load applied to your telephone line after you plug the system into the wall jack. The telephone line for your premises does not operate correctly if the total ringer load exceeds the capabilities of the telephone company's Central Office (CO) equipment. If too many ringers connect to the line, there may not be enough energy to ring your system. If the ringer load exceeds the system's capabilities, you can have problems dialing telephone numbers.

For more information about the total REN permitted for your telephone line, contact your local telephone company. However, as a guideline, a total REN of five should support normal operation of your equipment.

If your system equipment causes harm to the telephone network, the telephone company can temporarily discontinue your service. The telephone company can ask you to disconnect the equipment from the network until the problem is corrected and you are sure the equipment is working correctly. If possible, the telephone company notifies you before they disconnect the equipment. You are notified of your right to file a complaint with the FCC.

Your telephone company can make changes in its facilities, equipment, operations, or procedures that can affect the correct operation of your equipment. If the telephone company does make changes, they give you advance notice. With advance notice, it is possible for you to make arrangements to maintain uninterrupted service.

If you experience trouble with your system equipment, contact your authorized distributor or service center.

You cannot use the equipment on public coin service provided by the telephone company. Connection to party line service is subject to state tariffs. Contact the state public utility commission, public service commission, or corporation commission for information.

The equipment can provide access to interstate providers of operator services through the use of Equal Access codes. Failure to provide Equal Access capabilities is a violation of the Telephone Operator Consumer Services Improvement Act of 1990 and Part 68 of the FCC Rules.

Hearing aid compatibility

All proprietary telephones used with the system meet with the requirements of FCC Part 68 Rule 68.316 for hearing aid compatibility.

FCC compliance: Registered equipment for Direct Inward Dial calls

Equipment registered for Direct Inward Dial (DID) calls must provide proper answer supervision. Failure to meet this requirement is a violation of part 68 of the FCC's rules.

The definition of correct answer supervision is as follows:

- DID equipment returns answer supervision to the Central Office when DID calls are:
  - — answered by the called telephone
  - — answered by the attendant
  - — routed to a recorded announcement that can be administered by the user

— routed to a dial prompt

- DID equipment returns answer supervision on all DID calls forwarded to the Central Office. Exceptions are permitted if a call is not answered, a busy tone is received, or a reorder tone is received.

Radio and TV interference

The system complies with Part 15 of the FCC rules in the United States of America. Operation is subject to the following two conditions:

1. The system must not cause harmful interference.
2. The system must accept any interference received, including interference that can cause undesirable operation.

You can determine the presence of interference by placing a telephone call while monitoring. If the system causes interference to radio or television reception, try to correct the interference by moving the receiving TV or radio antenna if this can be done safely. Then move the TV or radio in relation to the telephone equipment.

If necessary, ask a qualified radio or television technician or supplier for additional information. You can refer to the document "How to Identify and Resolve Radio-TV Interference", prepared by the Federal Communications Commission. This document is available from:

U.S. Government Printing Office
Washington DC 20402

Notice for Canadian installations

Industry Canada uses a label to identify certified equipment. Certification indicates that the equipment meets certain operations, safety, and protection requirements for telecommunications networks. Industry Canada does not guarantee that the equipment will operate to the user’s satisfaction.

The Load Number (LN) assigned to each terminal device is the percentage of the total load that can be connected to a telephone loop using the device. This number prevents overload. The termination on a loop can have any combination of devices, provided that the total of the Load Numbers does not exceed 100. An alphabetical suffix is also defined in the Load Number for the appropriate ringing type (A or B), if necessary. For example, LN = 20 A indicates a Load Number of 20 and an "A" type ringer.

Before you install any equipment, make sure that it can connect to the facilities of the local telecommunications company. Install the equipment using acceptable methods of connection. In some cases, a certified connector assembly (telephone extension cord) can extend the company’s inside wiring associated with a single line individual service. Understand that compliance with the above conditions does not always prevent degradation of service.

Repairs to certified equipment must be made by an authorized Canadian maintenance facility designated by the supplier. If you make repairs or modifications to this equipment, or if the equipment malfunctions, the telephone company can ask you to disconnect the equipment.

Make sure that the electrical ground connections of the power utility, telephone lines, and internal metallic water pipe system, if present, connect together. This precaution is for the users' protection, and is very important in rural areas.

DANGER
DANGER OF ELECTRIC SHOCK
The system frame ground of each unit must be tied to a reliable building ground reference.

DANGER
DANGER OF ELECTRIC SHOCK
Do not attempt to make electrical ground connections yourself. Contact your local electrical inspection authority or electrician to make electrical ground connections.

Radio and TV interference

The system does not exceed Class A limits for radio noise emissions from digital apparatus, as set out in the radio interference regulations of Industry Canada (ICES-003).

Canadian and US network connections

Table 2 "Network connection specifications" (page 31) contains information that must be given to the local telephone company when ordering standard network interface jacks for the system.

Table 2 "Network connection specifications" (page 31) includes columns for system port identification, Facility Interface Code (FIC), Service Order Code (SOC), Uniform Service Order Code (USOC) jack identification, and associated Nortel equipment part numbers.

Table 2
Network connection specifications

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
MTS/WATS					
2-Wire, LSA, L-S (2-Wire, Local Switched Access, Loop-Start)	02LS2	9.0F	2.7A	RJ21X CA21X*	NT8D14
2-Wire, LSA, G-S (2-Wire, Local Switched Access, Ground-Start)	02GS2	9.0F	2.7A	RJ21X CA21X*	NT8D14
2-Wire, LSA, R-B (2-Wire, Local Switched Access, Reverse-Battery)	02RV2-T	9.0F	0.0B	RJ21X CA21X*	NT8D14
1.544 Mbps OSI, SF	04DU9-BN	6.0P	N/A	RJ48 CA48*	NTRB21
1.544 Mbps OSI, SF	04DU9-KN	6.0P	N/A	RJ48 CA48*	NTRB21
Analog PL facilities					
8-port OPX line	OL13C	9.0F	N/A	RJ21X	NT1R20
E&M TIE Trunk (TIE line, lossless, 2-wire type 1 E&M)	TL11M	9.0F	N/A	RJ2EX CA2EX*	NT8D15
* RJ with CA for Canada					

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
E&M 4-Wire DRTT (TIE line, lossless, dial repeating, 4-wire type 1 E&M)	TL31M	9.0F	N/A	RJ2GX CA2GX*	NT8D15
E&M 4-Wire DRTT (TIE line, lossless, dial repeating, 4-wire type 2 E&M)	TL32M	9.0F	N/A	RJ2HX CA2HX*	NT8D15
Digital					
1.544 Mbps superframe	04DU9-BN	6.0P	N/A	N/A	NT5D12
1.544 Mbps extended superframe	04DU9-KN	6.0P	N/A	N/A	NT5D12
* RJ with CA for Canada					

Notice for International installations

If there is not enough planning or technical information available for your country of operation, contact your regional distributor or authority.

European compliance information

The system meets the following European technical regulations: CTR 1, CTR 2, CTR 3, CTR 4, CTR 6, CTR 10, CTR 12, CTR 13, CTR 15, CTR 17, CTR 22, CTR 24, and the I-ETS 300 131.

Supported interfaces

Analog interfaces are approved based on national or European specifications. Digital interfaces are approved based on European specifications.

Safety specifications

The system meets the following European safety specifications: EN 60825, EN 60950, and EN 41003.

Notice for Germany

Empfangen und Auspacken des

Communication Server 1000E

Dem Gerät sollte eine Teileliste beiliegen, die alle im Lieferumfang des Systems enthaltenen Teile auflistet. Vergleichen Sie diese Teileliste mit den erhaltenen Teilen. Sollte die Teileliste mit den erhaltenen Teilen nicht übereinstimmen, benachrichtigen Sie unverzüglich den Lieferungsagenten und Nortel. Alle mit dem System bestellten Optionen sind werkseitig installiert und nicht separat auf der Teileliste aufgelistet. Bewahren Sie die Versandkartons auf, um sie ggf. wiederverwenden zu können.

Hinweis: Falls die Versandkartons bei Empfang beschädigt sind, sollten Sie den Lieferungsagenten bitten, bei dem Auspacken und der Inspektion des Geräts anwesend zu sein.

1. Stellen Sie sicher, daß sich der Verpackungskarton in aufrechter Position befindet.
2. Schneiden Sie das Verpackungsklebeband vorsichtig mit einem Schneidemesser auf, und öffnen Sie dann den Karton.
3. Entfernen Sie die Kartonverpackung, das Schaumstoff-verpackungsmaterial und die schützende Plastikverpackung.
4. Heben Sie das Chassis vorsichtig aus dem Karton, und plazieren Sie es an dem gewünschten Aufstellungsort.

Richtlinien zum Aufstellen des Systems

Bei der Wahl des Systemstandorts empfiehlt es sich, folgende Punkte in Betracht zu ziehen:

1. Stabilität. Stellen Sie das System in einem Bereich auf, der vor übermäßigen Bewegungen und Erschütterungen geschützt ist.
2. Sicherheit. Installieren Sie das System im Hinblick auf Sicherheit. Sorgen Sie dafür, daß Kabel und Drähte den Zugang nicht behindern.
3. Zugang. Stellen Sie das System so auf, daß es problemlos gewartet werden kann. Bei Wartungsarbeiten ist Zugang zur Vorder- und Rückseite des Systems erforderlich.
4. Betriebsumgebung. Stellen Sie das System in einem Bereich auf, an dem es Hitze, Staub, Rauch und elektrostatischer Entladung (ESE) nicht ausgesetzt ist.
5. Kühlung. Lassen Sie Platz für eine ausreichende Luftzirkulation zur Kühlung. Stellen Sie sicher, daß vor und hinter dem System mindestens 10 cm Freiraum gelassen wird. (Zusätzliche Richtlinien zur Kühlung des Gerätes finden Sie im nächsten Abschnitt.)

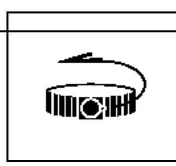
Kühlen des Gehäuses

Es ist äußerst wichtig, daß alle Geräte eines Systems sachgemäß gekühlt werden. Die Eingangslufttemperatur der Systemkomponenten muß im allgemeinen unter 50° C (122° F) liegen. Interne, durch Gleichstrom betriebene Ventilatoren kühlen die Laufwerke und Module des Systems ab. Die Übergangsmodule an der Rückseite des Chassis werden durch natürliche Konvektion gekühlt. Um eine ausreichende Kühlung zu gewährleisten, sollten Sie:

- Vor und hinter dem System mindestens 10 cm Freiraum lassen.
- Sicherstellen, daß die Verkleidungen aufgesetzt, alle vorderen und rückwärtigen Schlitze gefüllt und alle Öffnung abgedeckt sind.
- Alle nicht verwendeten Modulschlitze abdecken.

Bei der Installation des Systems in einer bestimmten Betriebsumgebung sollten die technischen Daten zur Betriebsumgebung der Systemkomponenten beachtet werden. Zum Beispiel: Bei Umgebungstemperaturen über 50° C (122° F) wird der Betrieb von Disketten- und Festplattenlaufwerken nicht mehr zuverlässig. Im Falle eines Gerätes, das in einem Gehäuse installiert ist, sollten Sie beachten, daß die interne Umgebungstemperatur unter Umständen über die maximal mögliche, externe Umgebungstemperatur ansteigen kann.

ESE und Sicherheit



ESE-ANTISTATIKBAND VERWENDEN

Nortel empfiehlt, bei allen Installations- oder Aufrüstarbeiten am System ein Antistatikband und eine ableitende Schaumstoffunterlage zu verwenden. Elektronische Komponenten, wie z.B. Plattenlaufwerke, Platinen und Speichermodule, können gegen ESE äußerst empfindlich sein. Nach dem Entfernen des Bauteils aus dem System oder aus der Schutzhülle wird das Bauteil flach auf eine geerdete und statikfreie Oberfläche gelegt, und im Falle einer Platine mit der Komponentenseite nach oben. Das Bauteil nicht auf der Oberfläche hin und her bewegen.

Ist kein ESE-Arbeitsplatz verfügbar, so können ESE-Gefahren durch das Tragen eines Antistatikbands (in Elektronik-Fachgeschäften erhältlich) vermieden werden. Dabei ist ein Ende des Bandes um das Handgelenk zu legen. Das Erdungsende (normalerweise ein Stück Kupferfolie oder eine Krokodilklemme) an einer elektrischen Masseverbindung anschließen. Hierbei kann es sich um ein Stück Metall handeln, das direkt zur Erde führt (z.B. ein unbeschichtetes Metallrohr) oder ein Metallteil eines geerdeten, elektrischen Gerätes. Ein elektrisches Gerät ist geerdet, wenn es einen dreistiftigen Schukostecker besitzt, der in eine Schuko-Steckdose gesteckt wird. Das System selbst kann nicht als Masseverbindung verwendet werden, weil es bei allen Arbeiten vom Netz getrennt wird.

DANGER
WARNUNG
Vor dem Ausführen dieser Verfahren ist die Stromzufuhr des Systems auszuschalten und das System vom Stromnetz zu trennen. Wenn der Strom vor dem Öffnen des Systems nicht ausgeschaltet wird, besteht die Gefahr von Körperverletzungen und Beschädigungen des Gerätes. Im Gerät sind gefährliche Spannungen, Strom und Hochenergie vorhanden. An den Anschlußpunkten der Betriebsschalter können gefährliche Spannungen anliegen, auch wenn sich der Schalter in der ausgeschalteten Position befindet. Das System darf nicht bei abgenommener Gehäuseabdeckung betrieben werden. Vor dem Einschalten des Systems ist die Gehäuseabdeckung stets anzubringen.

Sicherheits- und Betriebsnormen

Diese Systeme entsprechen den Sicherheits- und Betriebsnormen, die für einzelne Geräteteile gelten. Es ist jedoch möglich, dieses Produkt mit anderen Einzelteilen zusammen zu verwenden, die ein System ergeben, welches nicht den Systemrichtlinien entspricht. Da Nortel nicht voraussehen kann, welche Geräte mit diesem Gehäuse verwendet werden oder wie dieses Gehäuse verwendet wird, sind der Systemintegrator und der Installateur völlig dafür verantwortlich, daß das gesamte fertiggestellte System den Sicherheitsanforderungen von UL/CSA/VDE sowie den EMI/HFI-Emissionsgrenzen entspricht.

Vorsichtshinweise zur Lithium-Batterie

Dieses System enthält Lithium-Batterien.

CAUTION
VORSICHT
Bei einem inkorrekten Auswechseln der Lithium-Batterien besteht Explosionsgefahr. Wechseln Sie die Batterien nur mit dem gleichen oder einem gleichwertigen Batterietyp, der von dem Hersteller empfohlen ist, aus. Entsorgen Sie gebrauchte Batterien gemäß den Herstelleranweisungen.

CAUTION
VORSICHT
Bitte nehmen Sie vor Ort keine Wartung bzw. Austausch der Lithium-Batterien selber vor. Um die Batterien sachgemäß warten oder auswechseln zu lassen, setzen Sie sich mit Ihrem Nortel Servicevertreter in Verbindung.

Installation in ein 19-Zoll-Rack

Um das Gerät in ein Rack einzubauen, gehen Sie folgendermaßen vor:

CAUTION
VORSICHT
Befestigen Sie das Chassis nicht oben am Rack. Ein kopflastiges Rack kann Umkippen und Geräte beschädigen sowie Personal verletzen.

Um Verletzungen von Personen oder Beschädigungen der Geräte zu vermeiden sollten folgende Schritte von zwei Personen ausgeführt werden.

1. Schieben Sie das Chassis vorne in das Rack.
2. Befestigen Sie das Chassis mit Schrauben. (Um Genaueres über die hierzu empfohlenen Schraubenarten zu erfahren, wenden Sie sich bitte an den Hersteller des Racks.)
3. Stellen Sie sicher daß der Netzschalter (ON/OFF oder EIN/AUS) am Chassis auf OFF (O) gestellt ist. Ist Ihr System mit einem Spannungswahlschalter versehen, so stellen Sie den Schalter auf die Ihrem Standort gemäße Betriebsspannung.
4. Stecken Sie das Sockelende des Chassisnetzkabels in die Netzsteckbuchse an der Rückseite des Chassis.
5. Installieren Sie alle Kommunikationskabel.
6. Stecken Sie alle Netzkabel in eine geerdete, gegen Spannungsspitzen geschützte Schuko-Steckdose.
7. Um den Netzstrom einzuschalten, stellen Sie den Netzschalter (ON/OFF) an der Rückseite des Chassis auf ON (1). Die normale Startroutine des Systems erfolgt, und das System ist dann einsatzbereit.

DANGER
WARNUNG
Vor Wartungsarbeiten am Chassis ist das Netzkabel vom Stromnetz zu trennen, um die Gefahr eines elektrischen Schlages oder andere mögliche Gefahren zu reduzieren.

Data network planning for VoIP

Contents

This section contains information on the following topics:

"Introduction" (page 37)

"Data network planning for VoIP" (page 37)

"100BaseTx IP connectivity" (page 40)

Introduction

WARNING
Before a CS 1000E 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, see *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

The data network's infrastructure, engineering, and configuration are critical to achieve satisfactory IP Telephony voice quality. A technical understanding of data networking and Voice over IP (VoIP) is essential for optimal performance of the CS 1000E system.

See *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for detailed information about network requirements. These requirements are critical to the system Quality of Service (QoS).

Data network planning for VoIP

Consider the following when planning the network:

- system network requirements (for ELAN and TLAN subnets)

- basic data network requirements for Call Server to Media Gateway connections, including the following:
  - jitter
  - delay
  - bandwidth
  - LAN recommendations
- basic data network requirements for IP Phones
  - bandwidth
- power requirements for IP Phones

Evaluating the existing data infrastructure

Evaluate the existing data infrastructures (LAN and WAN) to confirm their suitability for VoIP deployment. In some cases, VoIP deployment requires additional bandwidth, improved performance and QoS, and greater availability.

To evaluate voice performance requirements, review such items as device inventory, network design, and baseline information on network performance. Links and devices must have sufficient capacity to support additional voice traffic. You may need to upgrade links that have high peak or busy hour utilization.

When assessing the environment, target devices with the following characteristics:

- high CPU utilization
- high backplane utilization
- high memory utilization
- queuing drops
- buffer misses for additional inspection
- potential upgrade

Peak utilization characteristics in the baseline are valuable in determining potential voice quality issues.

To evaluate requirements for the VoIP network, review network topology, feature capabilities, and protocol implementations. Measure redundancy capabilities of the network against availability goals with the network design recommended for VoIP.

Evaluate the overall network capacity to ensure that the network meets overall capacity requirements. Overall capacity requirements must not impact existing network and application requirements. Evaluate the network baseline in terms of the impact on VoIP requirements.

To ensure that both VoIP and existing network requirements are met, it can be necessary to add one or more of the following:

- memory
- bandwidth
- features

Planning deployment of a CS 1000E system on a data network

To deploy the CS 1000E system on a data network, consider the following data networking details and see Converging the Data Network with VoIP Fundamentals (NN43001-260):

- VoIP technology
  - H.323 protocols
  - VoIP concepts and protocols
  - RTP
  - Codecs including G.711 and G.729
- data network architecture
  - TCP/IP
  - IP subnetting
  - routing protocols including EIGRP, OSPF, RIP, and BGP
- data services and peripherals
  - DNS
  - DHCP
  - TFTP
  - Web server
  - QoS

QoS planning

An IP network must be engineered and provisioned to achieve high voice quality performance. It is necessary to implement QoS policies network-wide to ensure that voice packets receive consistent and proper treatment as they travel across the network.

IP networks that treat all packets identically are called "best-effort networks". In a best-effort network, traffic can experience varying amounts of delay, jitter, and loss at any time. This can produce speech breakup, speech clipping, pops and clicks, and echo. A best-effort network does not guarantee that bandwidth is available at any given time. Use QoS mechanisms to ensure bandwidth is available at all times, and to maintain consistent, acceptable levels of loss, delay, and jitter.

For planning details for QoS, see *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Core network planning

CS 1000E IP Telephony network design consists of two networks:

1. ELAN (Embedded LAN) subnet
2. TLAN (Voice LAN) subnet

The ELAN (Embedded LAN) subnet, isolates critical telephony signaling between the Call Server and the other components. The TLAN (Telephony LAN) subnet, carries telephony, voice, and signaling traffic, and connects to the customer network and the rest of the world.

100BaseTx IP connectivity

Between the Call Server and Media Gateway, the CS 1000E supports 100BaseTx IP connectivity or campus data network connectivity. Campus data network connectivity is provided through ELAN and Layer 2 switches.

To satisfy voice quality requirements, adhere to applicable engineering guidelines. See *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for details. Contact the local Data Administrator to obtain specific IP information.

Campus network system requirements

The following campus network system requirements are necessary:

- The ELAN subnet and the TLAN subnet must be separate.
- ELAN subnet applications must be on the same subnet. This includes the Voice Gateway Media Cards, which must be on the same ELAN subnet.
- Voice Gateway Media Cards in the same node must be on the same TLAN subnet.
- Use of the VLAN concept is a practical way to maintain the same subnet for remote locations.

See *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for information on basic data network and LAN requirements for Call Server to Media Gateway connections, including the following:

- Packet Delay Variation (PDV) jitter buffer
- bandwidth planning
- LAN recommendations for Excellent Voice Quality
- monitoring IP link voice quality of service
- basic data network requirements for IP Phones
  - — bandwidth requirements
  - — bandwidth planning

Media conversion devices

Third-party media conversion devices can extend the range of the 100BaseTx and convert it to fiber. Use caution when extending the length of cable used with a media converter. Do not exceed the specified round-trip delay parameters.

System architecture

Contents

"Main components" (page 43)

"CS 1000E Call Server" (page 47)

"Media Gateway" (page 57)

"Media Gateway Expander" (page 61)

"Media Gateway 1000E" (page 62)

"Signaling Server" (page 71)

"Terminal Server" (page 76)

"Layer 2 switch" (page 78)

"Power over LAN (optional)" (page 79)

"Telephones" (page 79)

"Component dimensions" (page 80)

Main components

CS 1000E systems can be configured for either Standard Availability or High Availability (system redundancy).

Figure 2 "CP PM CS 1000E Standard Availability" (page 44) shows the typical main components of a Standard Availability CP PM equipped CS 1000E solution.

Figure 3 "CP PM CS 1000E High Availability" (page 45)shows the typical main components of a High Availability CP PM equipped CS 1000E solution.

Figure 4 "CP PIV CS 1000E High Availability" (page 46) shows the typical main components of a High Availability CP PIV equipped CS 1000E solution.

ATTENTION

CP PII or CP PIV equipped CS 1000E cannot be configured for Standard Availability. CS 1000E systems equipped with a CP PM processor can be configured for Standard Availability or upgraded to High Availability with an additional CP PM processor and software package 410 HIGH_AVAIL HIGH AVAILABILITY. The remainder of this chapter discusses each component in further detail.

Figure 2
CP PM CS 1000E Standard Availability

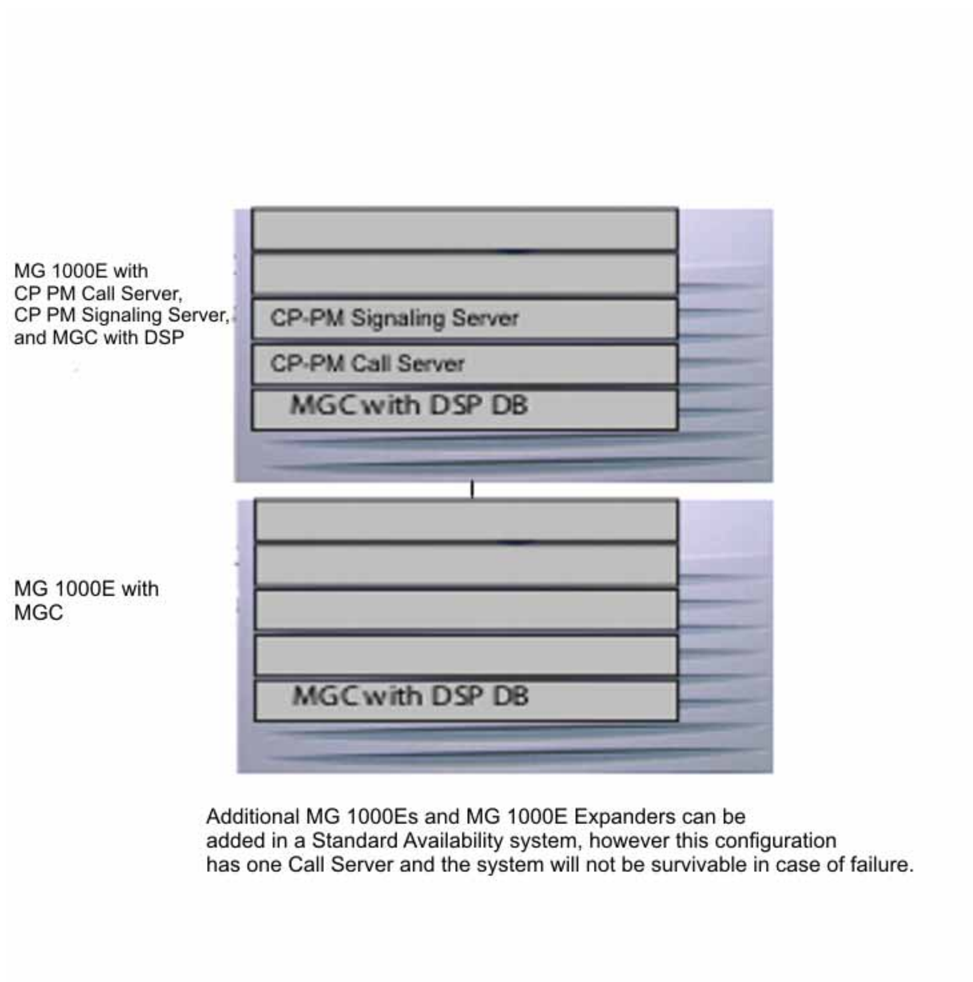


Figure 3
CP PM CS 1000E High Availability

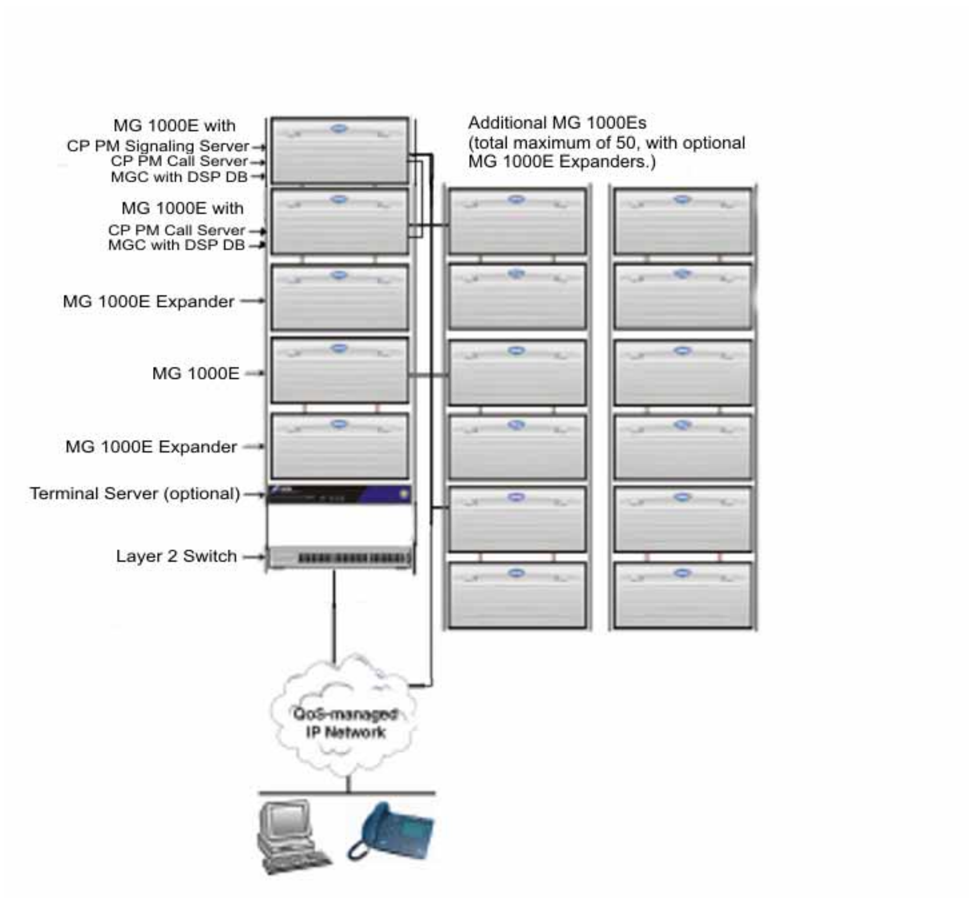
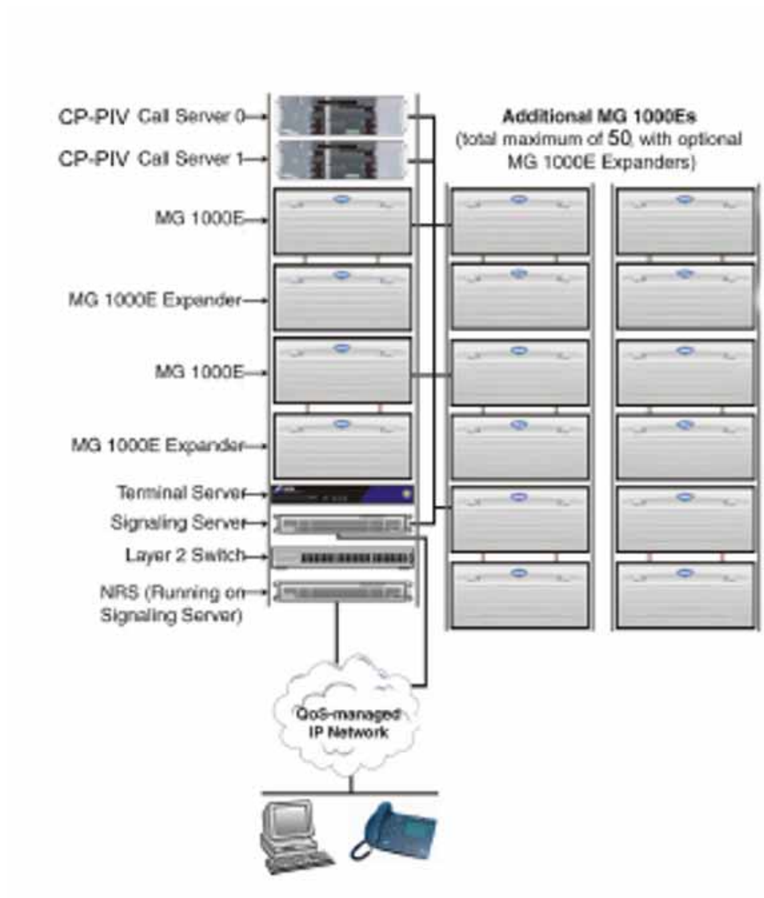


Figure 4
CP PIV CS 1000E High Availability



A typical CS 1000E solution is composed of a Call Server, Media Gateway, and Signaling Server. Additional components such as Layer 2 switches and Terminal Servers can also be equipped if required.

- The **Call Server** provides call processing capability. It includes a Standard Availability or High Availability CS 1000E Call Server. (see "CS 1000E Call Server" (page 47))
- The **MG 1000E** platform provides the CS 1000E system with IPE card slots for connection of telephones and trunks. CP PIV and CP PM Call Servers can support 1 to 50 Media Gateway 1000Es (MG 1000E) and optional MG 1000E Expanders. CP PII systems can support 1 to 30 MG 1000Es (see "Media Gateway 1000E" (page 62)).
- The **Signaling Server** provides the CS 1000E system with SIP/H.323 signaling between components. Signaling Servers (total number

required depends on capacity and survivability levels). (see "Signaling Server" (page 206))

- The **Layer 2 switch** provides the CS 1000E system with additional ports to transmit data packets to devices interconnected by Ethernet to the ELAN or TLAN subnets (see "Layer 2 switch" (page 78)).
- The **Terminal Server** provides the CS 1000E system with standard serial ports for applications and maintenance. MRV Terminal Server (only required in a CS 1000E system with a CP PIV call processor and SSC-based Media Gateways). (see "Terminal Server" (page 76)).

CS 1000E Call Server

Main role

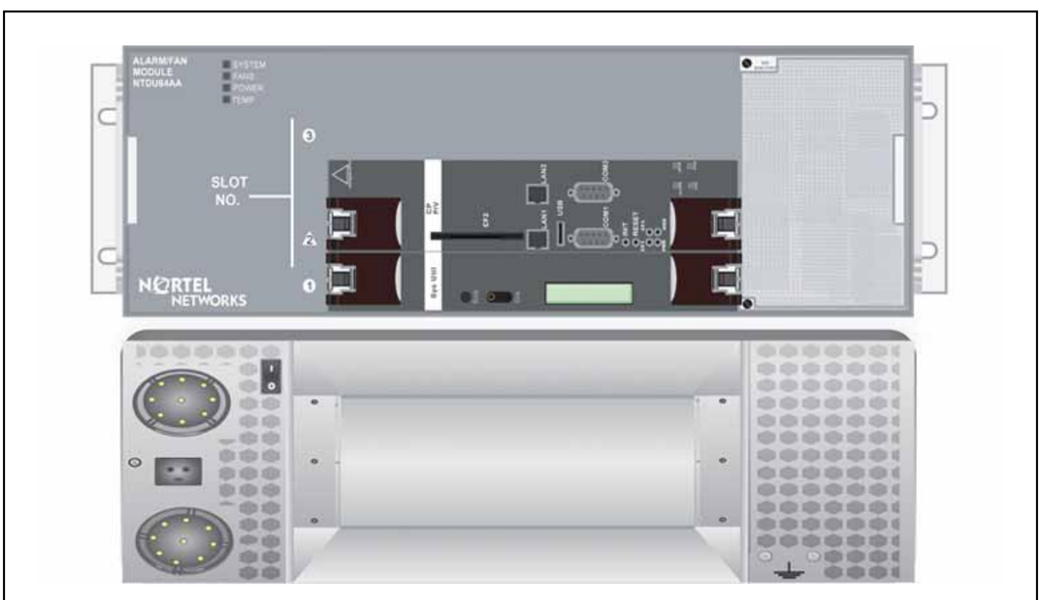
The CS 1000E Call Server serves as the call processor for the CS 1000E system. The Call Server can be either CP PII, CP PIV or CP PM based. CP PM Call Servers can be employed in a standalone or redundant configuration.

The CS 1000E system software is based on the core software of the CS 1000M Large System.

Physical description of a CP PIV Call Server

Figure 5 "CS 1000E CP PIV Call Server (front and rear)" (page 47) shows the front (without cover) and rear of one Call Server.

Figure 5
CS 1000E CP PIV Call Server (front and rear)



CP PIV Call Server hardware components

Similar to the set of core circuit cards used in CS 1000M Large System, each CP PIV Call Server contains the following:

- CP PIV Call Processor card
- System Utility card

In addition, each Call Server is equipped with the following modules:

- Power supply module
- Alarm/fan module

CP PIV Call Processor card

The CP PIV Call Processor card (NT4N39AA) is the main processor for the Call Server, controlling all call processing and telephony services. It also provides the system memory required to store operating software and customer data.

The CP PIV Call Processor card provides the following connectors:

- The **Com 1 port** connects to an IP-based Terminal Server, which provides standard serial ports for system maintenance and third-party applications (for more information, see "Terminal Server" (page 76)). The Com 1 port can also be directly connected to a system terminal for system access.
- The **Com 2 port** can be used as an additional RS-232 port (for system maintenance only).
- The **LAN 1 Ethernet port** connects the Call Server to the Embedded LAN (ELAN) subnet through an ELAN Layer 2 switch to provide IP connections between the Call Server, Signaling Servers, and MG 1000Es. The port is a 10/100/1000MB auto-negotiate port.
- The **LAN 2 Ethernet port** connects Call Server 0 to Call Server 1 over a 1 Gbps auto negotiating high speed pipe to provide communication and database synchronization.
- The **USB port** is not supported by the CS 1000E system and cannot be used.

System Utility card

The System Utility card (NT4N48) provides auxiliary functions for the Call Server.

The minimum vintage for the System Utility card with CS 1000E is NT4N48BA.

System Utility card functions include:

- LCD display for system diagnostics

- interface to the Call Server alarm monitor functions
- Core-selection DIP switches to specify Call Server 0 or Call Server 1
- software security device holder

The software security device enables the activation of features assigned to the CS 1000E system. The security device for a CS 1000E Call Server is similar to the one used on a CS 1000M Large System

Filler Blank

The filler blank covers over the disk carrier slot used in the older CP PII-based system. The blank supports the blue LEDs that illuminate the Nortel Logo.

Power supply module

The AC power supply module (NTDU65) is the main power source for the Call Server and is field-replaceable.

Alarm/fan module

The alarm/fan module (NTDU64) provides fans for cooling the Call Server and provides status LEDs indicating the status of Call Server components. The alarm/fan module is field-replaceable.

Physical description of a CP PM Call Server

Figure 6 "CS 1000E CP PM Call Server (front)" (page 50) shows the front (without cover) of a NTDU14 chassis with a MGC card and CP PM card.

Figure 7 "Option 11C cabinet upgraded to CS 1000E CP PM Call Server" (page 51) shows the front (without cover) of an Option 11C cabinet upgraded with MGC and CP PM cards.

Figure 6
CS 1000E CP PM Call Server (front)

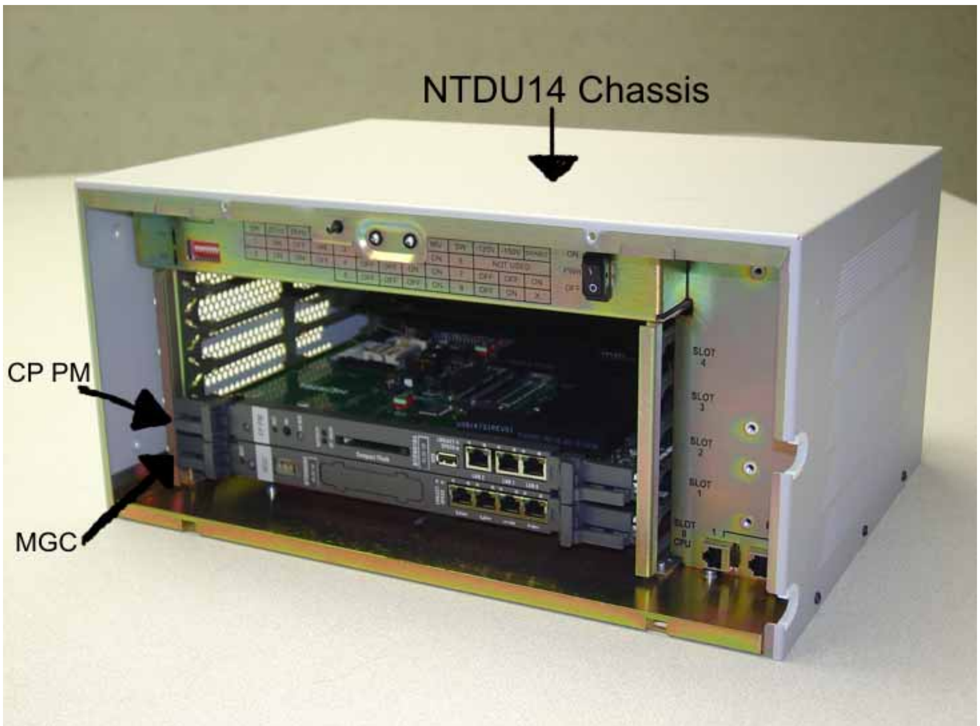


Figure 7
Option 11C cabinet upgraded to CS 1000E CP PM Call Server

CP PM Call Server hardware components

Each CP PM Call Server requires the following hardware:

- CP PM supported chassis:
  - — Option 11C cabinet (except for slot 0, slot 5 and slot 6)
  - — Option 11C Mini chassis (except for slot 0 and slot 4)
  - — Option 11C Mini expander chassis

— MG 1000E main chassis (except for slot 0)

— MG 1000E expander chassis

- CP PM Call Processor card
- MGC card

Call Server chassis

"Media Gateway" (page 57) provides information regarding the NTDU14 chassis. For information about upgrading Option 11C equipment to be used as a Call Server, see*Communication Server 1000E Upgrades (NN43041-458)*.

CP PM card

The CP PM card (NTDW61) is the main processor for the call server. The CP PM offers similar features to that of the CP PIV/CP PII processor, but uses an IPE slot form factor, allowing for a CS 1000E product with only a single MG 1000E chassis.

Figure 8 "CP PM card NTDW61" (page 53) shows the CP PM faceplate and CP PM circuit card.

Figure 8
CP PM card NTDW61

CP PM hardware includes the following components and features:

- Intel Pentium M processor
- Integrated Intel 855MGE GMCH/Intel ICH-4 controller chipset
- Two Compact Flash sockets: 1 GB fixed media disk (FMD) on the card and a hot swap removable media disk (RMD) accessible on the faceplate
- DDR RAM expandable up to 2 GB

- Three Ethernet ports (TLAN, ELAN, HSP):
- One USB 2.0 port, for future use
- Security device, housed on board

CP PM System Utility functionality

CP PM Call Server solution does not require an external System Utility (SUTL) card

The CP PM provides similar SUTL functionality without requiring any external hardware. Instead of setting a switch to indicate the CPU side information, side settings are configured from the overlays or from the installation program. SUTL functionality provides a redundancy LED (CPU display), and security device.

Media Gateway Controller (MGC) card

The MGC card occupies the system controller slot 0 in the Media Gateway 1000E chassis.

The MGC card (NTDW60) provides a gateway controller for IP Media Gateways in a CS 1000E system. The MGC only functions as a gateway controller under control of a CS 1000E (CP PIV or CP PM) call server. It is an upgrade over the Small System Controller (SSC) in this application. SSC can still be equipped with a CP PII or CP PIV Call Server. There can also be a mix of SSC and MGC with CP PII or CP PIV Call Servers.

The MGC card supports up to 128 DSP channels (equivalent of 4 Media Cards) with the two expansion sites to accommodate digital signal processor daughterboards. 96-port DSP daugherboard NTDW64 and 32-port DSP daughterboard NTDW62 can be installed on the MGC. The DSP daughterboards support VoIP voice gateway resources on the MGC, reducing the need for separate Voice Gateway Media Cards.

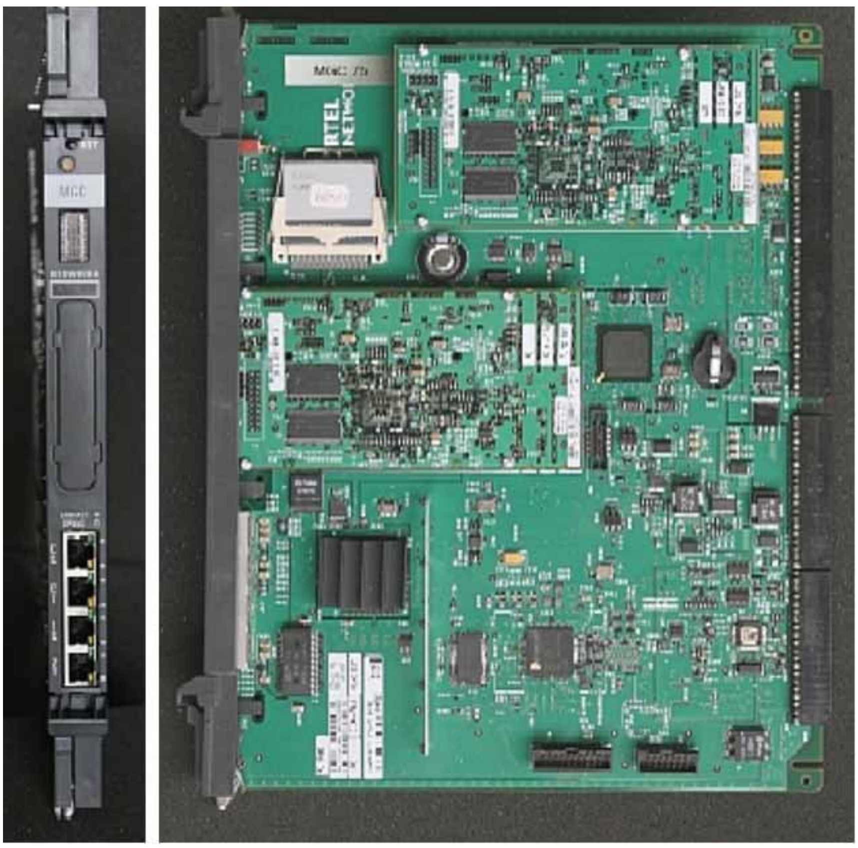
The MGC DSP daughterboard security feature provides an infrastructure to allow endpoints capable of SRTP/SRTCP to engage in secure media exchanges. The media security feature can be configured by the administrator or, optionally, by the end user. This feature provides for the exchange of cryptographic material needed by the SRTP-capable endpoints to secure media streams originating from those endpoints.

For more information about Media Security or SRTP, see *Security Management (NN43001-604)*.

DSP daugherboards include voice gateway (VGW) application; they do not include the Terminal Proxy Server (TPS) application. DSP daughterboards cannot be used for load sharing of IP Phones from Signaling Servers, Voice Gateway Media Cards, or as backup TPS in case of failures.

Figure 9 "Media Gateway Controller card (NTDW60)" (page 55) shows the MGC faceplate and MGC circuit card (with two DSP daughterboards installed).

Figure 9
Media Gateway Controller card (NTDW60)



The MGC card (without expansion daughterboards) includes the following components and features:

- Arm processor
- 128 MB RAM
- 4 MB boot flash
- Internal Compact Flash (CF) card mounted on the card. It appears to the software as a standard ATA hard drive
- Embedded Ethernet switch
- Six 100 BaseT Ethernet ports for connection to external networking equipment
- Four character LED display on the faceplate

- Two PCI Telephony Mezzanine Card form factor sites for system expansion
- Real time clock (RTC)
- Backplane interface
- Three serial data interface (SDI) ports

Figure 10
NTDW63 MGC adapter for Option 11C

When installing an MGC in an Option11C cabinet, a NTDW63 adapter is required to provide ports for Ethernet and clock reference.

> **ATTENTION**
> The MGC is a gateway controller that can replace the SSC in an MG 1000E. It also reduces the need for separate Voice Gateway Media Cards with the use of onboard DSP daughterboards. MGC-based MG 1000E supports PRI/PRI2/DTI/DTI2 trunks, BRI trunks, D-channels, and clock controllers, eliminating the need for a separate MG 1000T peer gateway. The MGC remote SDI feature reduces the need for separate Terminal Servers.

Functional description

The Call Servers provide the following functionality:

- provide main source of call processing
- process all voice and data connections
- control telephony services
- control circuit cards installed in MG 1000Es
- provide resources for system administration and user database maintenance

Operating parameters

The CS 1000E can be equipped as Standard Availability (single Call Server) or High Availability (dual Call Server) (Core 0 and Core 1) to provide a fully redundant system.

Core 0 and Core 1 can operate in redundant mode over the HSP with software package 410 HIGH_AVAIL HIGH AVAILABILITY: one runs the system while the other runs in a warm standby mode, ready to take over system control if the active Call Server fails.

The system configuration and user database are synchronized between the active and inactive Call Servers. This lets the inactive Call Server assume call processing in the event of failure of the active Call Server.

The Call Server uses a proprietary protocol to control the MG 1000Es. This proprietary protocol is similar to industry-standard Media Gateway Control Protocol (MGCP) or H.248 Gateways.

CP PIV and CP PM Call Servers can control up to 50 MG 1000Es. CP PII Call Servers can control up to 30 MG 1000Es.

The Call Servers provide connectivity to telephony devices using IP signaling through MG 1000Es rather than by direct physical connections.

The CS 1000E system supports lineside T1 (NT5D14) and lineside E1 (NT5D34) cards. For further information about T1/E1 lineside cards, see *Circuit Card Reference (NN43001-311)*

Media Gateway

Main role

The Media Gateway houses IPE circuit cards and connectors for access to the Main Distribution Frame. For more information see "Media Gateway 1000E" (page 62) for details.

The MG 1000B uses the same base Media Gateway hardware as the MG 1000E. (For more information, see *Branch Office Installation and Commissioning (NN43001-314)* .

Physical description

Figure 11 "Media Gateway" (page 58) shows the Media Gateway (NTDU14).

Figure 11
Media Gateway

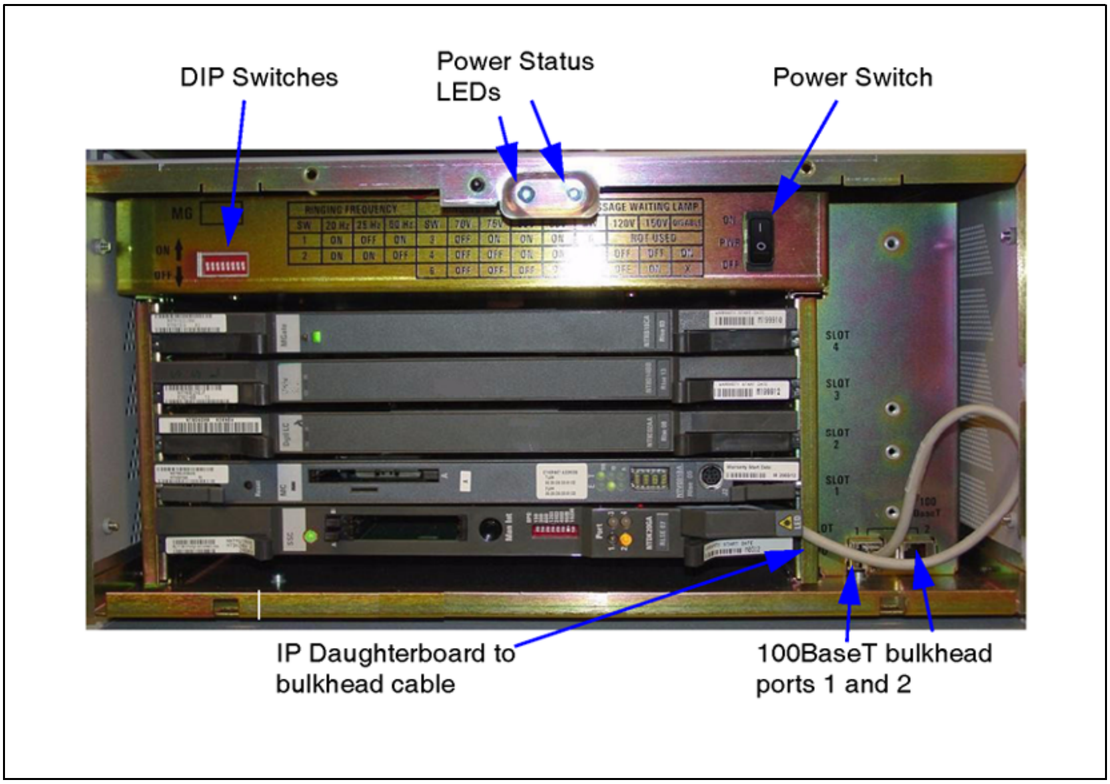
Hardware components

Front components

Figure 12 "Front components in the Media Gateway (NTDU14)" (page 59) shows the Media Gateway with the front cover removed. Note the following:

- The DIP switches telephone ringing voltages, ringing frequencies, and message waiting voltages.
- The 100BaseT bulkhead ports 1 and 2 provide MGC or SSC daughterboard ports with connections to rear bulkhead ports.

Figure 12
Front components in the Media Gateway (NTDU14)



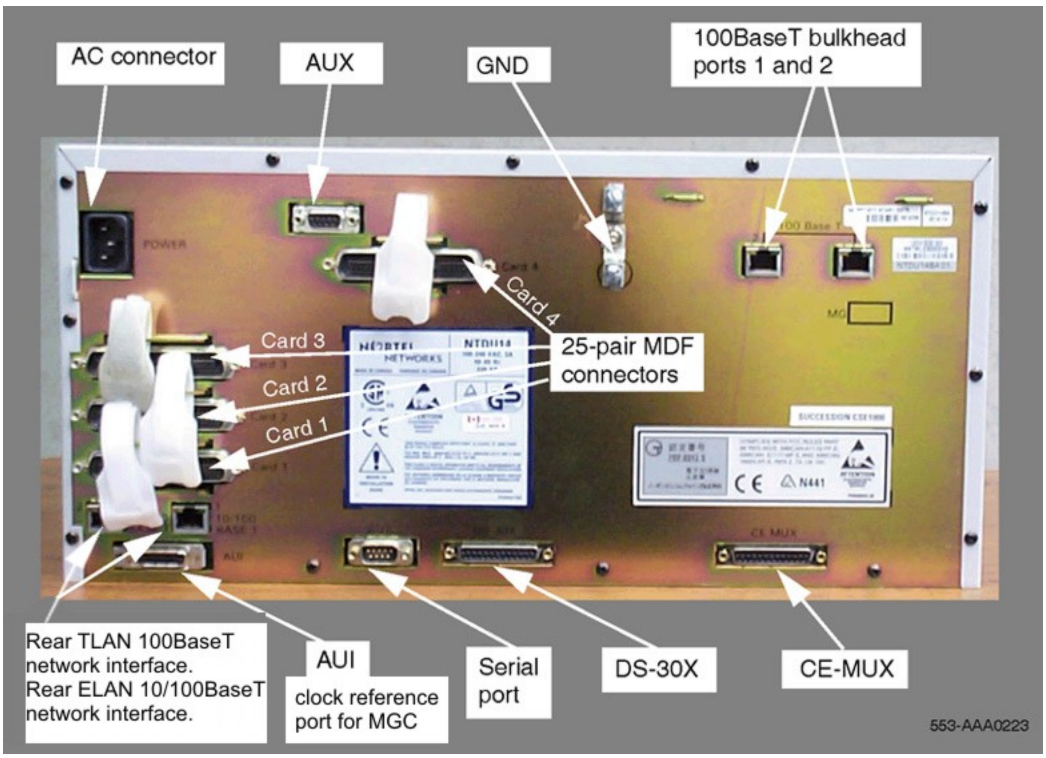
Rear components

Figure 13 "Rear components in the Media Gateway" (page 60) shows the rear components on the Media Gateway. Note the following:

- The AC power cord connector provides AC connection to the Media Gateway.
- AUX extends Power Failure Transfer Unit (PFTU) signals to the Main Distribution Frame (MDF).
- GND is used for ground cable termination.
- 100BaseT bulkhead ports 1 and 2 provide connections from IP daughterboard ports on the MGC or SSC card to other system components.
- The Attachment Unit Interface (AUI) is used with SSC cards that require a Media Access Unit (MAU).
- The AUI is used with MGC cards as a clock reference. When the AUI connection is used with an MGC, Ethernet link and speed LEDs cannot function.
- The serial port connects to maintenance terminals.

- DS-30X and CE-MUX interconnect the Media Gateway to the Media Gateway Expander.
- 25-pair connectors extend the IPE card data to the MDF.

Figure 13
Rear components in the Media Gateway



Circuit cards

Each Media Gateway can house the following circuit cards:

- MGC card (see Figure 9 "Media Gateway Controller card (NTDW60)" (page 55)
- CP PM card (see Figure 8 "CP PM card NTDW61" (page 53)
- SSC card (not used in the Expander) (see Figure 17 "Small System Controller card (NTDK20)" (page 64) for card details).
- Voice Gateway Media Cards (see "Voice Gateway Media Card" (page 65).
- Intelligent Peripheral Equipment (IPE) cards (see "Media Gateway 1000E" (page 62) for specific cards supported on each Media Gateway type).

Media Gateway Expander

Main role

The Media Gateway Expander supports up to four circuit cards. A new feature of the Media Gateway Expander is support for CP PM CS, and CP PM SS directly. An MGC or SSC card in the corresponding MG 1000E controls each card in an Expander. Digital trunk interface packs are not supported in the Media Gateway Expander.

Physical description

Figure 14 "Media Gateway Expander (NTDU15)" (page 61) shows the Media Gateway Expander (NTDU15).

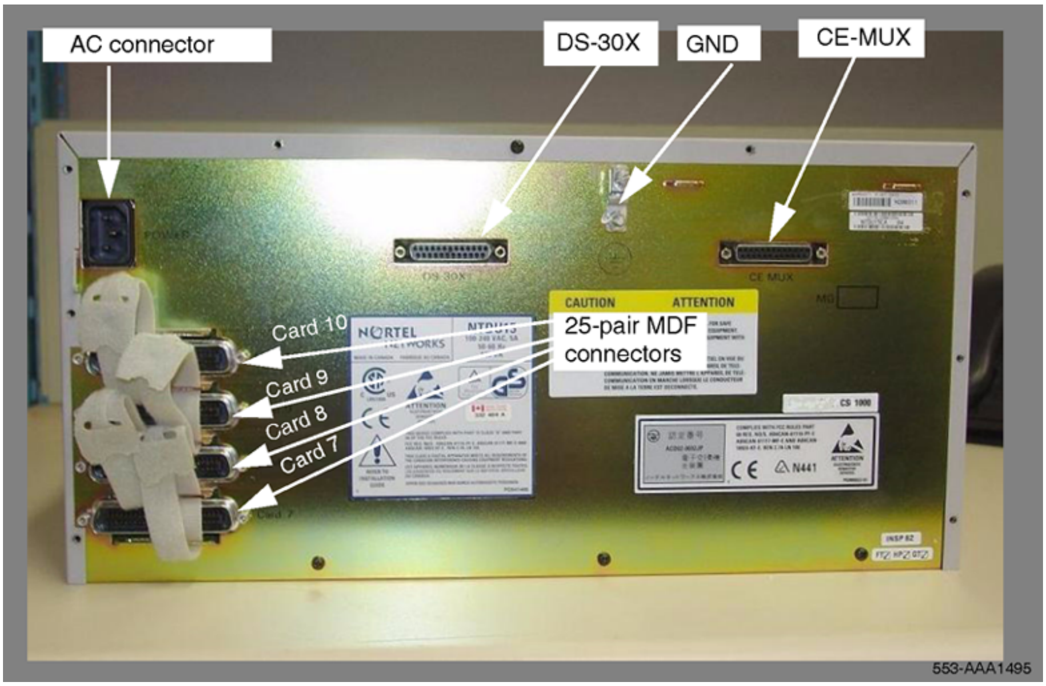
Figure 14
Media Gateway Expander (NTDU15)

Rear components

Figure 15 "Rear components in the Media Gateway Expander" (page 62) shows the rear components in the Expander. Note the following:

- The AC power cord connector provides an AC connection to the Expander.
- GND is used for ground cable termination.
- DS-30X and CE-MUX are used to interconnect the Media Gateway and the Expander.
- 25-pair connectors are used to extend IPE card data to the MDF.

Figure 15
Rear components in the Media Gateway Expander



Operating parameters

Each Media Gateway supports one optional Expander.

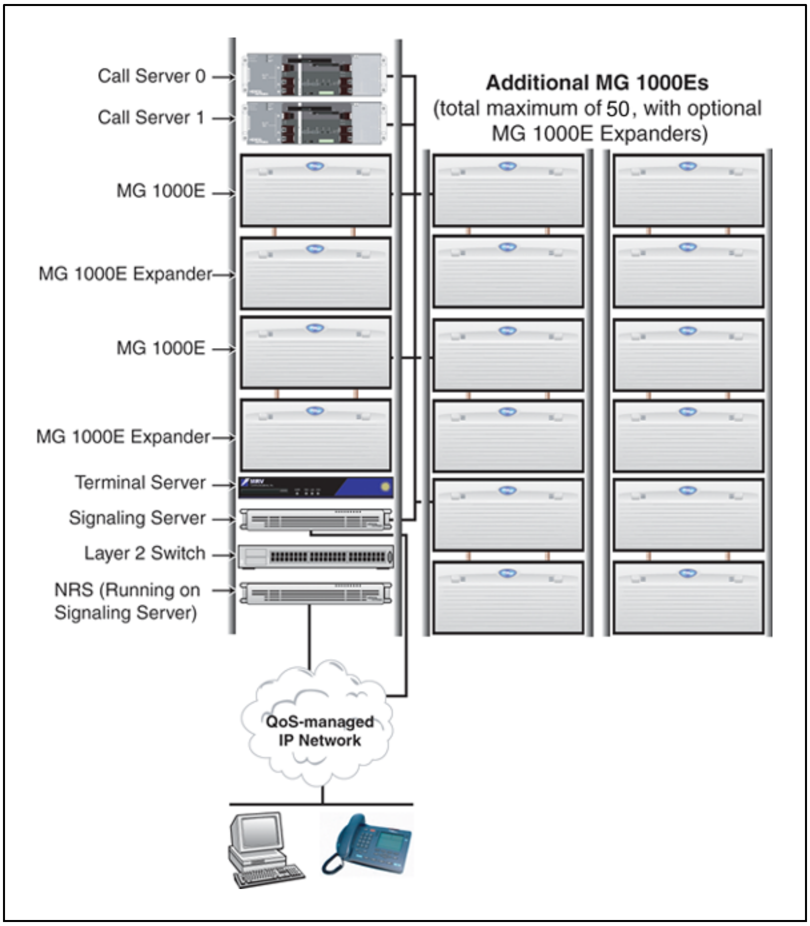
Media Gateway 1000E

Main role

The Media Gateway 1000E (MG 1000E) provides basic telephony media services, including tone detection, generation, and conference, to CS 1000E telephones. The Media Gateway 1000E with Media Gateway Controller supports digital trunk and PRI access to the PSTN and to other PBX systems. The MG 1000E also supports Nortel Integrated Applications, including Integrated Recorded Announcer. It can also provide connectivity for digital and analog (500/2500-type) telephones as well as analog trunks for telephone and fax.

Figure 16 "Media Gateway 1000Es" (page 63) shows the MG 1000Es controlled by the High Availability CP PIV based CS 1000E Call Server and connected to the IP network.

Figure 16
Media Gateway 1000Es



Hardware components

The MG 1000E houses an MGC or SSC card (SSC only supported with CP PII or CP PIV call processors) and contains four slots for IPE cards. Each MG 1000E supports an optional MG 1000E Expander through copper connections. For more details about the Media Gateway chassis, see "Media Gateway" (page 57) and "Media Gateway Expander" (page 61).

Small System Controller (SSC)

The SSC card (NTDK20) contains software that controls interface cards and application cards in the Media Gateway. The SSC card hardware resources provide 32 channels of conferencing and 60 channels of tone generation.

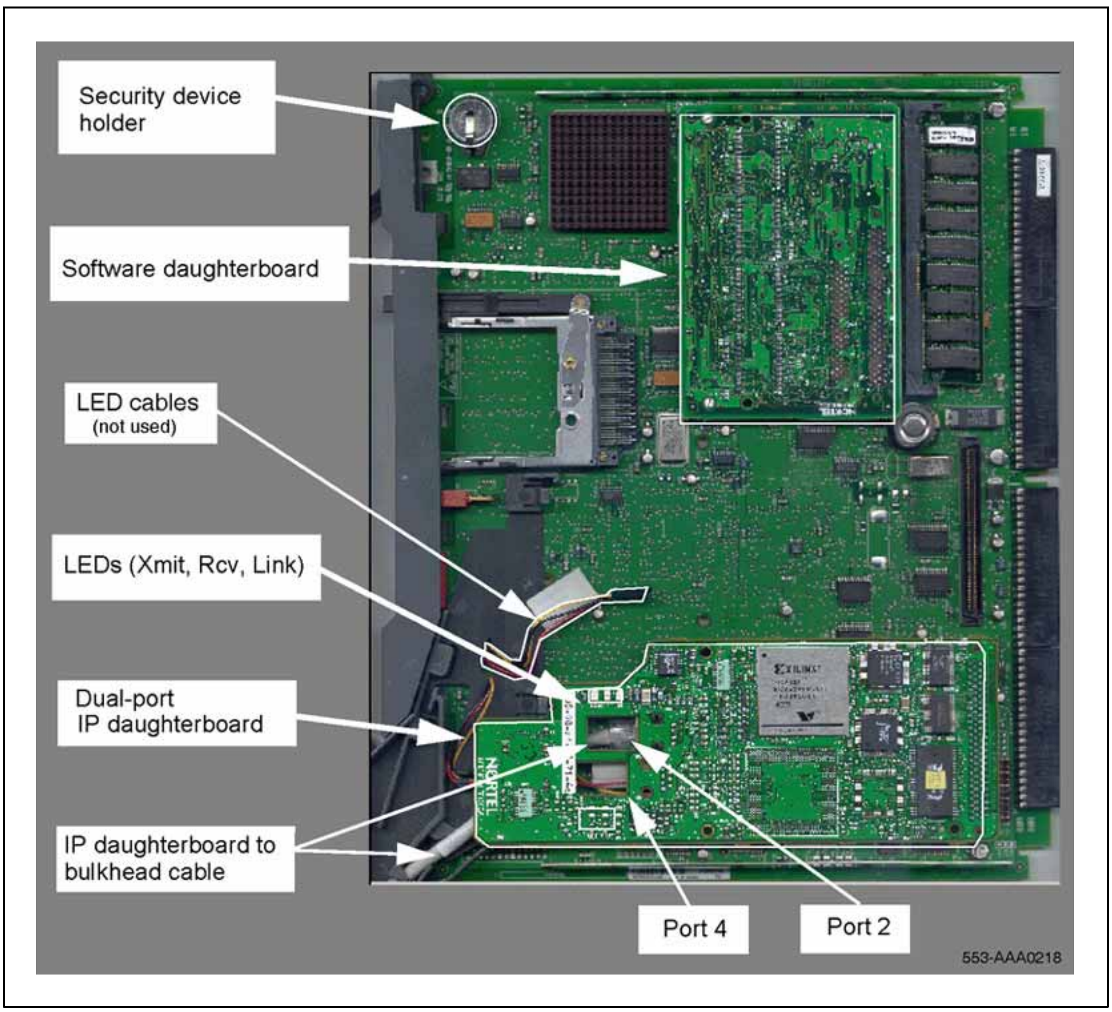
The SSC card is also equipped with IP daughter boards that provide 100BaseT IP interfaces to the ELAN subnet. Each IP daughterboard connected to the SSC also provides an additional 16 conference channels per port (16 channels with each single-port daughterboard and 32 channels with each dual-port daughterboard).

The SSC card also houses the software security device for the Media Gateway.

A specific type of security device is required for each type of Media Gateway. The SSC is not supported in an MG 1000E that is controlled with CP PM based CS 1000E. The SSC is only supported in CP PII or CP PIV based CS 1000E systems. There can be a mix of MG 1000Es with SSC and MGC cards in a CP PII and CP PIV based CS 1000E, but some features and capacities are reduced.

Figure 17 "Small System Controller card (NTDK20)" (page 64) shows the components in the SSC card.

Figure 17
Small System Controller card (NTDK20)



Security device

The security device on the MG 1000E SSC card is a generic security device that lets the MG 1000Es register with the CS 1000E Call Servers.

Control for the activation of features assigned to the CS 1000E system, including MG 1000Es, is provided by the security device on the System Utility card in the CP PIV Call Servers.

For more information about security devices, see *Security Management (NN43001-604)*.

Voice Gateway Media Card

Voice Gateway Media Cards provide interfaces that connect to the Telephony Local Area Network (TLAN) and Embedded Local Area Network (ELAN) subnets. Voice Gateway Media Card is also known as the ITG card or Media Card (MC). The Media Card can also run various applications. A Voice Gateway Media Card is any Media Card that runs the IP Line application. A Voice Gateway Media Card provides Digital Signal Processor (DSP) ports to translate between IP and TDM. Each Voice Gateway Media Card can provide 8, 24, or 32 DSP ports depending on the model installed.

Figure 18 "Media Card" (page 66) shows faceplate connectors and indicators on the 32-port line Media Card.

Figure 19 "Media Card 32S" (page 67) (MC32S) shows the faceplate connectors and circuit card of the 32-port secure Media Card.

Figure 18
Media Card

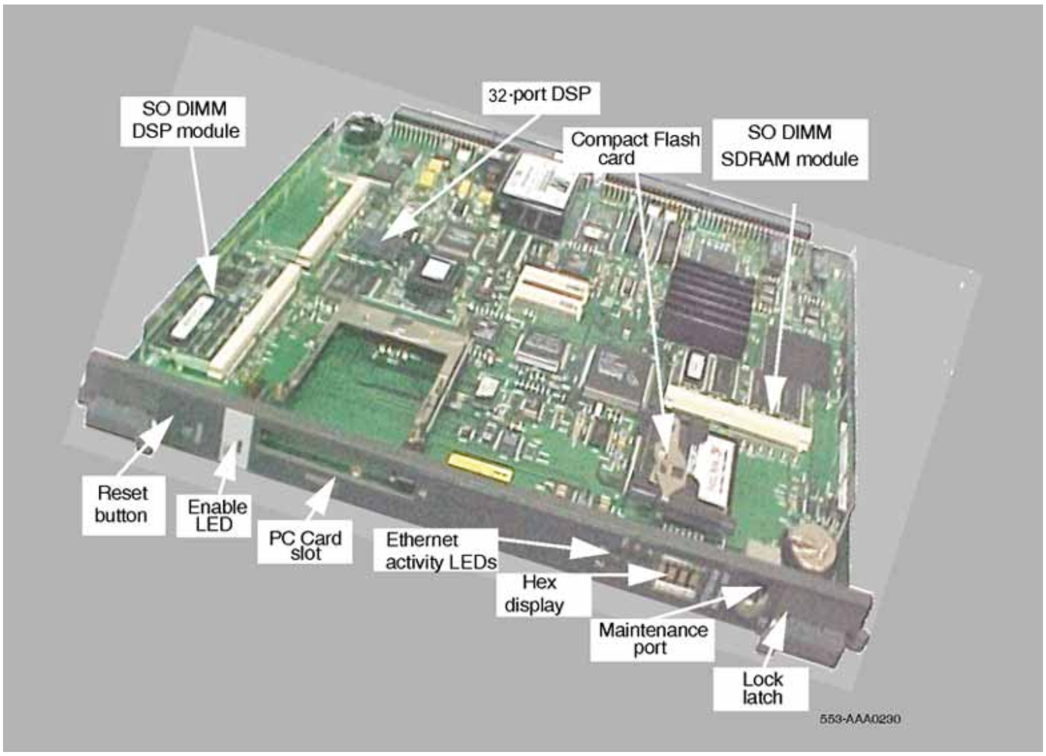


Figure 19
Media Card 32S

The MC32S provides 32 channels of IP-TDM connectivity between an IP device and a TDM device in the CS 1000 network. The MC32S is an IPE form factor card and can interwork with other voice gateway application cards such as the MGC, ITG-SA, and ITG-Pentium cards.

The MC32S Media Security feature provides an infrastructure to allow endpoints capable of SRTP/SRTCP to engage in secure media exchanges. The media security feature can be configured by the administrator or, optionally, by the end user. This feature provides for the exchange of cryptographic material needed by the SRTP-capable endpoints to secure media streams originating from those endpoints.

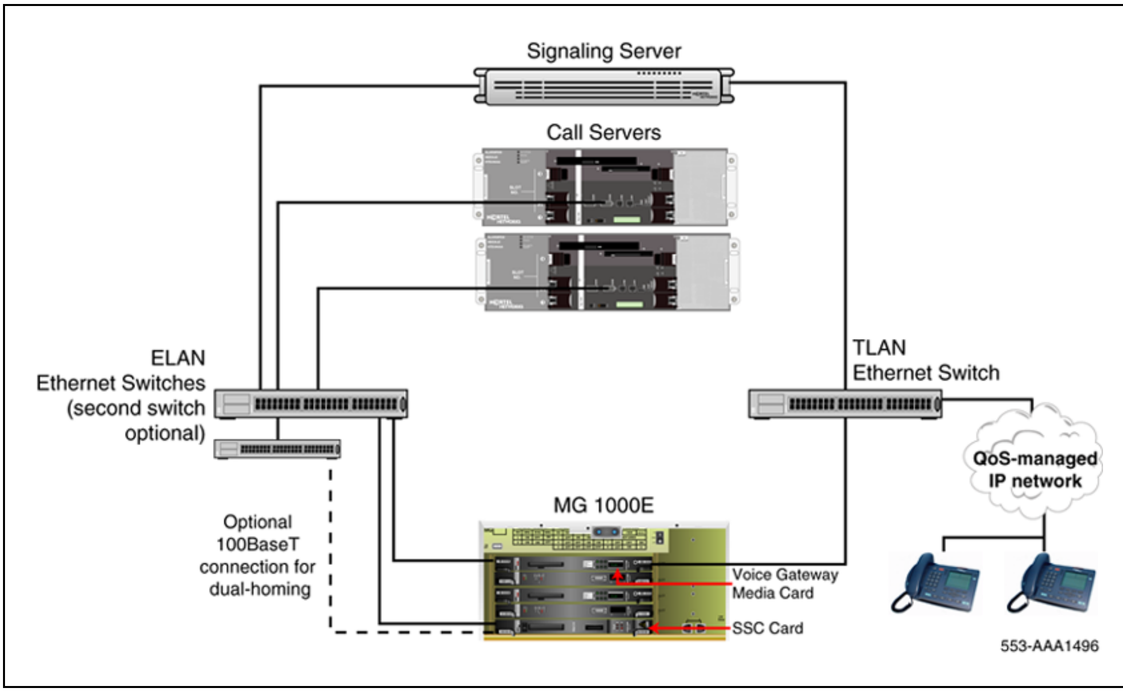
For more information about Media Security or SRTP, see *Security Management (NN43001-604)*.

For more information about Media Card features or the IP Line application, see *IP Line Fundamentals (NN43100-500)*.

Network connections

Figure 20 "Network connections on MG 1000E with SSC" (page 68) is a schematic representation of the typical network connections for one MG 1000E with SSC controlled by CP PIV Call Servers.

Figure 20
Network connections on MG 1000E with SSC



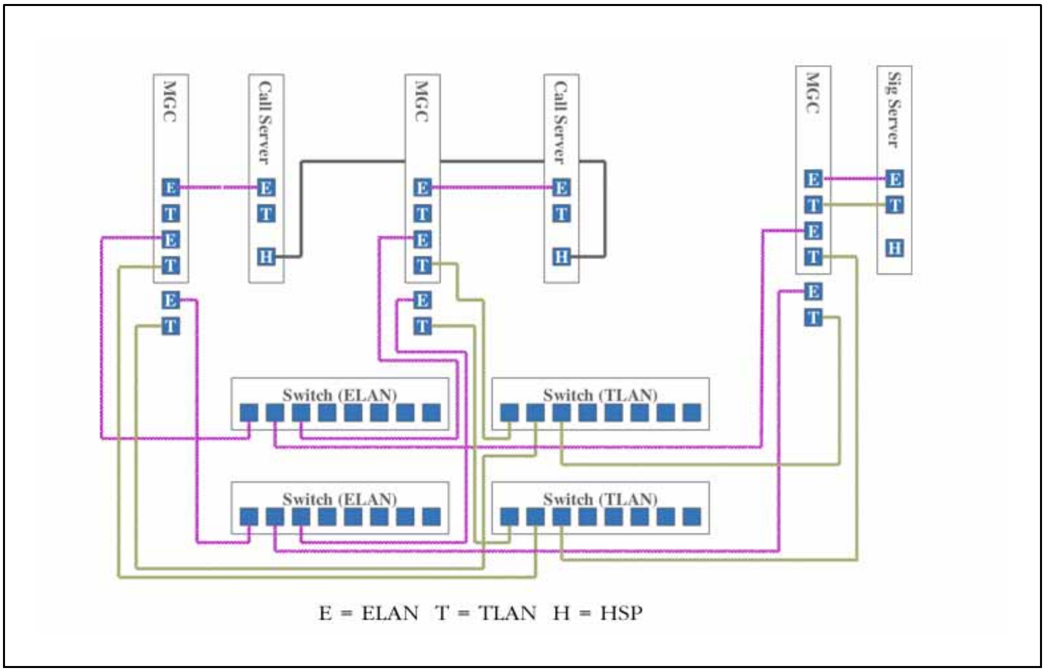
The dual-port IP daughterboard on the SSC card in the MG 1000E provides two 100BaseT ports for communication with the Call Server. In the basic configuration, Port 2 must connect to the ELAN through a Layer 2 or Layer 3 switch.

The ELAN of the MG 1000E can reside in a separate layer 3 subnet from that of the Call Server ELAN.

When connecting the MG 1000E to the ELAN through a Layer 3 switch the connection from the Call Server to the MG 1000E must have a round trip delay of less than 80 msec and have a packet loss of less than 0.5 % (0% recommended).

Figure 21 "Redundant network connections with MGC Dual Homing" (page 69) is a schematic representation of redundant network connections for MG 1000Es with MGC. The call servers, signaling servers, switches, and chassis can be any of the supported types.

Figure 21
Redundant network connections with MGC Dual Homing



The separate LAN subnets that connect the MG 1000E and the Call Server to the customer IP network are as follows:

- **ELAN** The ELAN subnet (100BaseT, full-duplex) is used to manage signaling traffic between the Call Server, Signaling Server, and MG 1000Es. The ELAN subnet isolates critical telephony signaling between the Call Servers and the other components.
- **TLAN** The TLAN subnet (100BaseT, full-duplex) is used to manage voice and signaling traffic. It connects the Signaling Server and Voice Gateway Media Cards to the Customer LAN. It also isolates the IP Telephony node interface from broadcast traffic.

The **HSP** (high speed pipe) is a 1000BaseT connection used to provide standby call server redundancy. The HSP provides connectivity for High Availability if two CP PM or CP PIV call servers are connected through the HSP, and the Campus Redundancy software package 410 HIGH_AVAIL HIGH AVAILABILITY has been purchased.

Functional description

The MG 1000E provides the following functionality:

- tones, conference, and digital media services (for example, Music and Recorded Announcement) to all phones
- support for CallPilot and Nortel Integrated Applications

- direct physical connections for analog (500/2500-type) phones, digital phones, and fax machines
- direct physical connections for analog trunks
- provides digital trunks to the PSTN and trunking to other PBX systems using E1, T1, and ISDN BRI circuit cards
- supports analog trunks
- supports Voice Gateway Media Cards for transcoding between IP and TDM
- supports the DECT application

Operating parameters

The MG 1000E operates under the direct control of the Call Server. Up to 50 MG 1000Es can be configured on the Call Server.

To allow IP Phones to access digital media services, the MG 1000Es must be equipped with Voice Gateway Media Cards.

The MG 1000E supports the following circuit cards and applications:

- Voice Gateway Media Cards: transcode between the IP network and digital circuit cards
- Service cards: provide services such as Music or Recorded Announcements (RAN)
- Analog interfaces to lines and trunks: support analog (500/2500-type) phones and fax, analog PSTN trunks, and external Music or RAN sources
- Analog trunk cards
- Digital line cards: support digital terminals, such as attendant consoles, M2000/M3900 series digital phones, and external systems that use digital line emulation, such as CallPilot Mini
- Digital PSTN Interface Cards, including E1, T1, and ISDN Basic Rate interfaces: provide access to PSTN
- CLASS Modem card (XCMC)
- DECT Mobility cards
- Nortel Integrated Applications, including:
  - — Integrated Conference Bridge
  - — Integrated Call Assistant
  - — Integrated Call Director
  - — Integrated Recorded Announcer
  - — Hospitality Integrated Voice Services

— MGate cards for CallPilot

— CallPilot IPE

Digital Trunks, PRI and BRI, and DECT Mobility Cards are only supported in MG 1000E with MGC.

Signaling Server

Main role

The Signaling Server provides SIP/H.323 signaling between components in a CS 1000E system.

The hardware platforms are ISP1100 Signaling Server, CP PM Signaling Server, and Commercial off-the-shelf (COTS) Signaling Servers. Available COTS servers are IBM x306m and HP DL320 G4 servers. The Signaling Server is used to run multiple applications, including:

- SIP/H.323 Signaling Gateways
- Terminal Proxy Server (TPS)
- Network Routing Service (NRS)
- Element Manager
- Application Server for Personal Directory, Callers List, and Redial List for UNIStim IP Phones

COTS servers can be configured for NRS on Linux, or NRS on VxWorks. COTS servers running NRS cannot run signaling server applications. COTS servers running NRS are not referred to as Signaling Servers.

VxWorks is supported on ISP 1100, CP PM and the 1U COTS platforms. Linux is supported on the 1U COTS platform.

NRS on VxWorks includes:

- H.323 Gatekeeper
- SIP Redirect Server
- NCS

NRS on Linux includes:

- H.323 Gatekeeper
- SIP Proxy Server
- NCS

For full details regarding Signaling Servers, refer to *Signaling Server Installation and Commissioning (NN43001-312)*

ISP1100 Signaling Server physical description

Figure 22 "ISP1100 Signaling Server front and rear" (page 72) shows the ISP1100 Signaling Server.

Figure 22
ISP1100 Signaling Server front and rear

ISP1100 Signaling Server hardware components

The front of the ISP1100 Signaling Server has the following components:

- A CD-ROM drive to load software files for the Signaling Server, Voice Gateway Media Cards, and IP Phones.
- A floppy disk drive if the CD-ROM is not bootable.
- A Maintenance port for a login session for Command Line Interface (CLI) management. (It does not provide system messages.)

Maintenance ports are on the front and back of the ISP1100 Signaling Server. Do not use both ports at once. The rear port is the recommended primary maintenance port.

The rear of the Signaling Server has the following components:

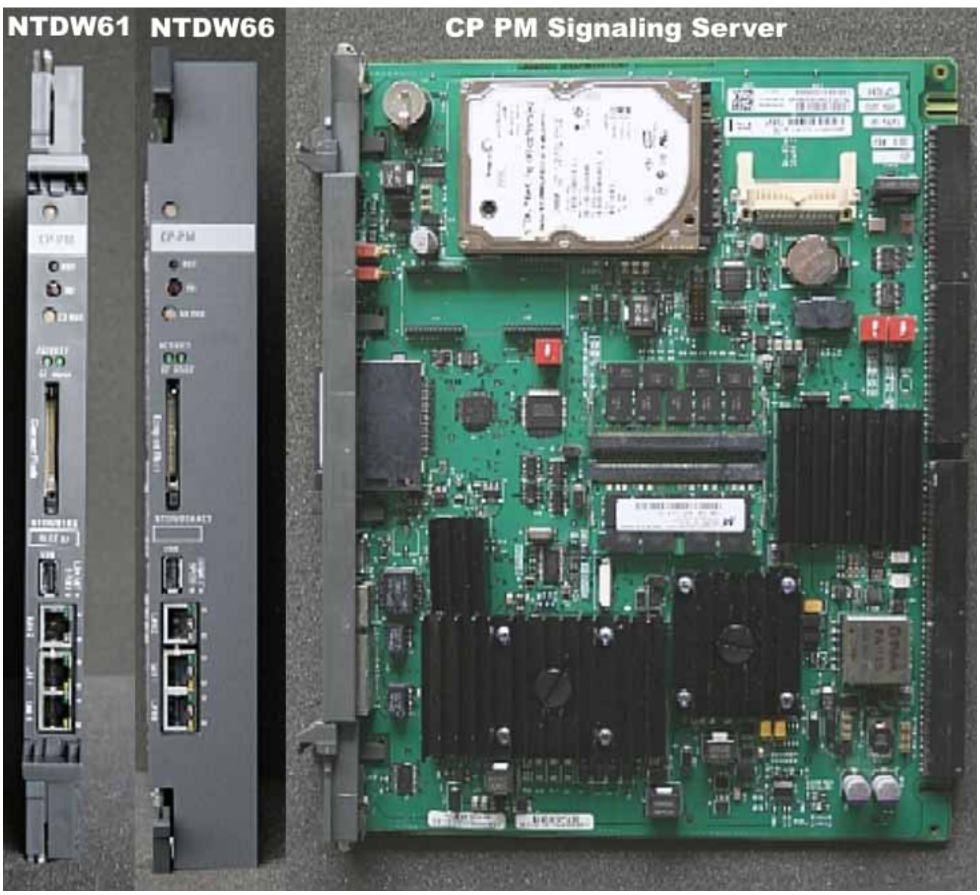
- The AC power cord connector provides an AC connection to the Signaling Server.
- The 100BaseT TLAN network interface is used for telephony signaling traffic.
- The 100BaseT ELAN network interface connects the Signaling Server to the Call Server and to the other CS 1000E components on the ELAN subnet.
- The rear maintenance port is the primary port for maintenance and administration terminals.

- The remaining ports are not used for any system functions. Do not plug any device into these ports.

CP PM Signaling Server physical description

Figure 23 "CP PM Signaling Server card and faceplate" (page 73) shows the CP PM Signaling Server. Two versions of the CP PM Signaling Server are marketed, each designed for use in specific chassis, however they are functionally identical. NTDW61 single slot CP PM card for MG 1000E, MG 1000B, or Option 11C cabinets. NTDW66 double wide faceplate CP PM card for CS 1000M Large System IPE cubes.

Figure 23
CP PM Signaling Server card and faceplate



CP PM Signaling Server hardware components

The CP PM Signaling Server hardware is based on the same board design as the CP PM Call Server. The CP PM Signaling Server uses a 40 GB fixed media device (hard drive), as opposed to the Compact Flash drive on the CP PM Call Server. The CP PM Signaling Server is populated with 1 GB of SDRAM. "CP PM card" (page 52) provides further details the CP PM hardware components.

COTS Signaling Server physical description

Commercial off-the-shelf Signaling Servers are multivendor servers based on 1U rack-mounted server chassis design.

Figure 24 "IBM x306m Signaling Server" (page 74) shows the IBM x306m Signaling Server.

Figure 25 "HP DL320 G4 Signaling Server" (page 75) shows the HP DL320 G4 Signaling Server.

Figure 24
IBM x306m Signaling Server

Figure 25
HP DL320 G4 Signaling Server

COTS Signaling Server hardware components

COTS Signaling Servers provide the following features:

- Intel Pentium 4 processor at 3.6 GHz
- One 80 GB SATA simple swap hard drive (2 front bays, 1 used)
- 2 GB PC2-4200 ECC DDR2 SDRAM (4 DIMM slots, 2 used)
- Two 10/100/1000BaseT Ethernet ports
- Four USB 2.0 ports (2 front, 2 back)
- One slim line CD-RW/DVD ROM drive
- One serial port

COTS servers do not have an INI button. COTS servers have a reset button on the front panel to perform cold starts. Software warm starts can be invoked through the command line interface (CLI) on the servers (reboot).

Software applications

The following software components operate on the Signaling Server:

- Terminal Proxy Server (TPS)
- SIP Gateway (Virtual Trunk)
- H.323 Gateway (Virtual Trunk)
- H.323 Gatekeeper
- Network Routing Service
  — SIP Redirect Server

- — SIP Registrar
- — Solid database component
- — NRS Manager

- CS 1000 Element Manager
- Application Server for the Personal Directory, Callers List, and Redial List features

Signaling server 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 descriptions of the function and engineering requirements of each element, see Table 33 "Elements in Signaling Server" (page 207). For detailed Signaling Server engineering rules and guidelines, see "Signaling Server algorithm" (page 235). For more information about H.323, SIP Trunking NRS and SIP Proxies see *IP Peer Networking Installation and Commissioning (NN43001-313)* and *Network Routing Service Installation and Commissioning (NN43001-564)*.

Functional description

The Signaling Server provides the following functionality:

- provides IP signaling between system components on the LAN
- enables the Call Server to communicate with IP Phones and MG 1000Ts
- supports key software components (see "Software applications" (page 75))

Operating parameters

The Signaling Server provides signaling interfaces to the IP network using software components that run on the VxWorks operating system.

The Signaling Server can be installed in a load-sharing, survivable configuration.

The total number of Signaling Servers that you require depends on the capacity and redundancy level that you require (see "Signaling Server algorithm" (page 235)).

Terminal Server

Main role

The MRV IR-8020M IP-based Terminal Server provides the Call Server with standard serial ports for applications and maintenance.

ATTENTION
A CS 1000E configured with MGC does not require a separate Terminal Server. The MGC provides serial ports for connectivity.

Physical description

Figure 26 "Terminal Server" (page 77) shows the Terminal Server.

Figure 26
Terminal Server

Hardware components

The MRV Terminal Server provides 20 console ports for modular RJ-45 connectors. It is also equipped with one RJ-45 10BaseT connection for network interface to the ELAN subnet and an internal modem to provide remote access.

Operating parameters

Traditionally, serial ports are used to connect terminals and modems to a system for system maintenance. As well, many third-party applications require serial port interfaces to connect to a PBX. Because the Call Server provides only two local serial ports for maintenance purposes, an IP-based Terminal Server is required to provide the necessary serial ports.

The Terminal Server provides standard serial ports for applications. These applications include billing systems that analyze Call Detail Recording (CDR) records, Site Event Buffers (SEB) that track fault conditions, and various legacy applications such as Property Management System (PMS) Interface and Intercept Computer applications. In addition, serial ports are used to connect system terminals for maintenance, modems for support staff, and printers for system output.

The Terminal Server is configured to automatically log in to the active Call Server at start-up. For this reason, each Call Server pair requires only one Terminal Server. Customers can configure up to 16 TTY ports for each Call Server pair.

The Terminal Server can be located anywhere on the ELAN subnet. However, if the Terminal Server is used to provide local connections to a Com port on the Call Server, it must be collocated with the system.

The Terminal Server can also be used as a central point to access and manage several devices through their serial ports.

> **ATTENTION**
> **IMPORTANT!**
> Currently, the CS 1000E only supports the MRV IR-8020M commercial Terminal Server.

Layer 2 switch

Main role

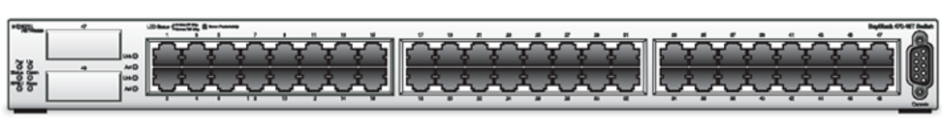
The Layer 2 switch transmits data packets to devices interconnected by Ethernet to the ELAN or TLAN subnets. The switch only directs data to the target device, rather than to all attached devices.

Physical description

ELAN Layer 2 switch

Figure 27 "ELAN Layer 2 switch (BayStack 470-48T)" (page 78) shows an example of an ELAN Layer 2 switch.

Figure 27
ELAN Layer 2 switch (BayStack 470-48T)



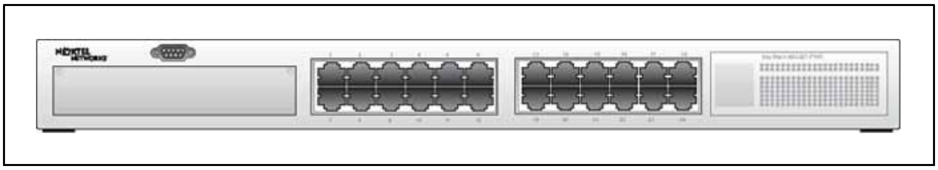
TLAN Layer 2 switch

To provide Layer 2 connections on the TLAN subnet, Nortel recommends the BayStack 460 Ethernet switch, which has embedded Power-over-LAN capabilities for powering IP Phones.

Optionally, other Power-over-LAN units can also be used to provide power to IP Phones.

Figure 28 "TLAN Layer 2 switch (BayStack 460)" (page 79) shows the BayStack 460 Layer 2 switch.

Figure 28
TLAN Layer 2 switch (BayStack 460)



Operating parameters

These components must be supplied by the customer. See *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for further 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 Uninterruptable Power Supply (UPS) for power backups. Using a Power over LAN unit eliminates the need to use a separate power transformer for each IP Phone.

See *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for further details.

Telephones

The CS 1000E system supports the following:

- IP Phones
  - — IP Phone 2001
  - — IP Phone 2002
  - — IP Phone 2004
  - — IP Phone 2007
  - — IP Phone 1120E
  - — IP Phone 1140E
  - — IP Phone 1150E
  - — IP Softphone 2050
  - — IP Audio Conference Phone 2033
  - — WLAN Handset 2210, 2211 and 2212
  - — IP Phone Key Expansion Module (KEM)
- analog (500/2500-type) telephones
- digital telephones

- attendant consoles
- DECT handsets
- 802.11 Wireless LAN terminals

Component dimensions

All rack mount components fit in 19-inch racks. Table 3 "Height dimension of CS 1000E components" (page 80) lists the height of each rack mount component. COTS Servers require the use of 4 post racks.

Option 11C cabinet is a supported CS 1000E enclosure. Option 11C cabinets are 25" high x 22" wide. They can be wall or floor mounted. Option 11C cabinets are not rack mountable.

Table 3
Height dimension of CS 1000E components

Component	Height
NTDU62 CP PIV Call Server	3 U
Signaling Server (ISP1100 and COTS)	1 U
NTDU14 Media Gateway	< 5 U
NTDU15 Media Gateway Expander	< 5 U
MRV Terminal Server	1 U
BayStack 460	< 2 U
BayStack 470	1 U
1 U = 4.4 cm (1-3/4 in.)	

The clearance in front of rack-mounted equipment is the same for all major components. For the Call Servers, Media Gateways, and Media Gateway Expanders, the distance from the mounting rails of the rack to the front of the bezel/door is 7.6 cm (3 in.).

Configuration options

Contents

"Introduction" (page 81)

"Option 1: Campus-distributed MG 1000Es" (page 82)

"Option 2: Campus Redundancy" (page 83)

"Option 3: Branch Office" (page 84)

"Option 4: Geographic Redundancy Survivable Media Gateway" (page 85)

Introduction

The IP-distributed architecture of the CS 1000E enables flexibility when it comes to component location. Given this flexibility, the CS 1000E offers many configuration options to support increased system redundancy.

The CS 1000E can be deployed in LAN and WAN environments. Most fall into one of the following categories:

- Multiple buildings in a campus
  - Campus-distributed MG 1000Es
  - Campus Redundancy
- Multiple sites
  - Central Call Server with Branch Office
  - Geographic Redundancy
  - Geographic Redundancy Survivable Media Gateway

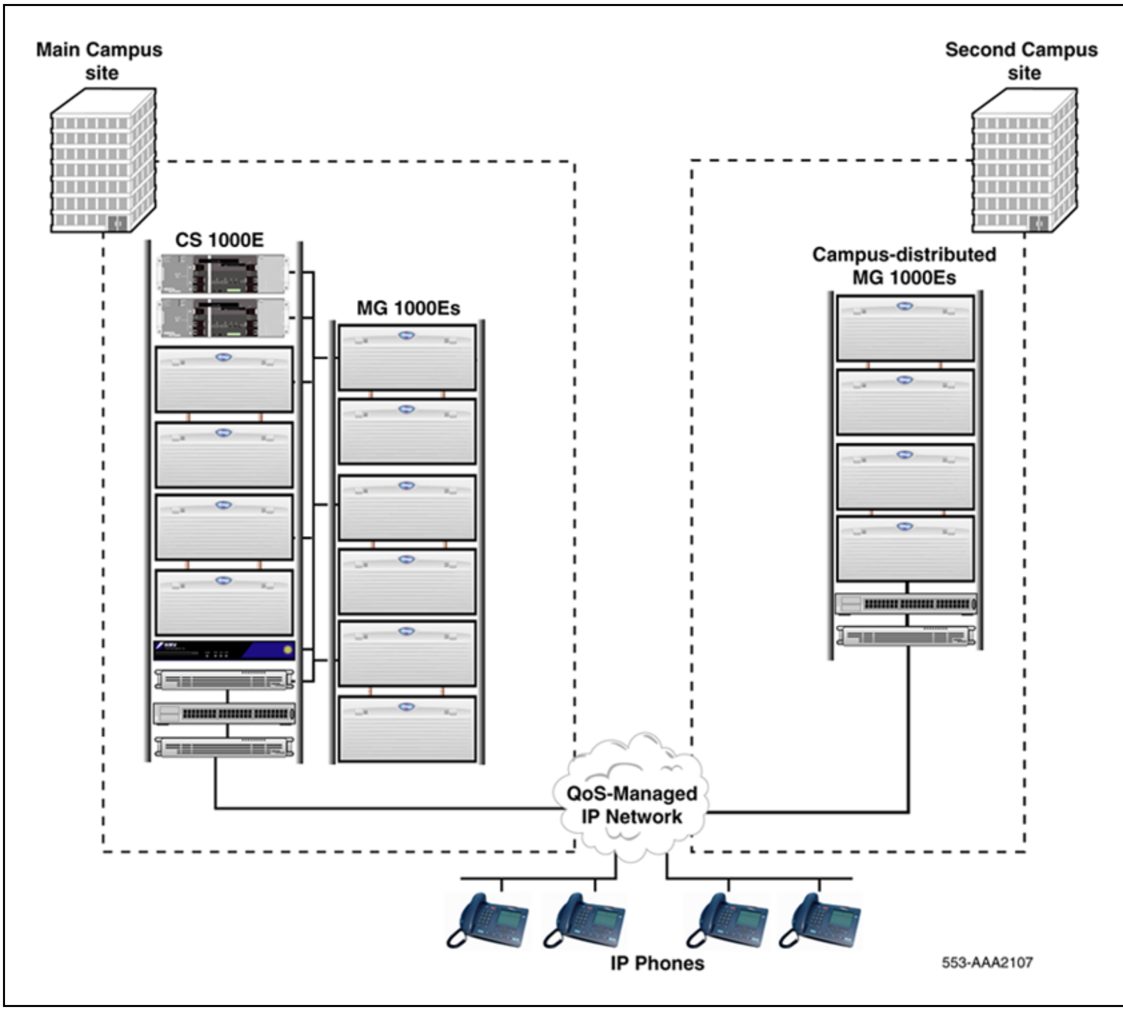
These configurations provide CS 1000E systems with many options for redundancy and reliability. Careful planning is required to determine the configuration that is appropriate for your needs.

The following sections describe each of these configuration options.

Option 1: Campus-distributed MG 1000Es

With multiple buildings in a campus, you can distribute MG 1000Es across a campus IP network. Figure 29 "Campus-distributed MG 1000Es" (page 82) shows MG 1000Es distributed across multiple buildings in a campus setting.

Figure 29
Campus-distributed MG 1000Es



In this configuration, a CS 1000E system is installed at the main site, and additional MG 1000Es and an optional Signaling Server are installed at a second campus site. All IP Phones are configured and managed centrally from the main site.

For details on the specific operating and network parameters for the MG1000E, see "Media Gateway 1000E" (page 62)

Modem traffic

The CS 1000E supports modem traffic in a campus-distributed network with the following characteristics:

- Media Card configuration:
  - G.711 codec
  - 20 msec packet size
- one-way delay less than 5 msec
- low packet loss

Performance degrades significantly with packet loss.

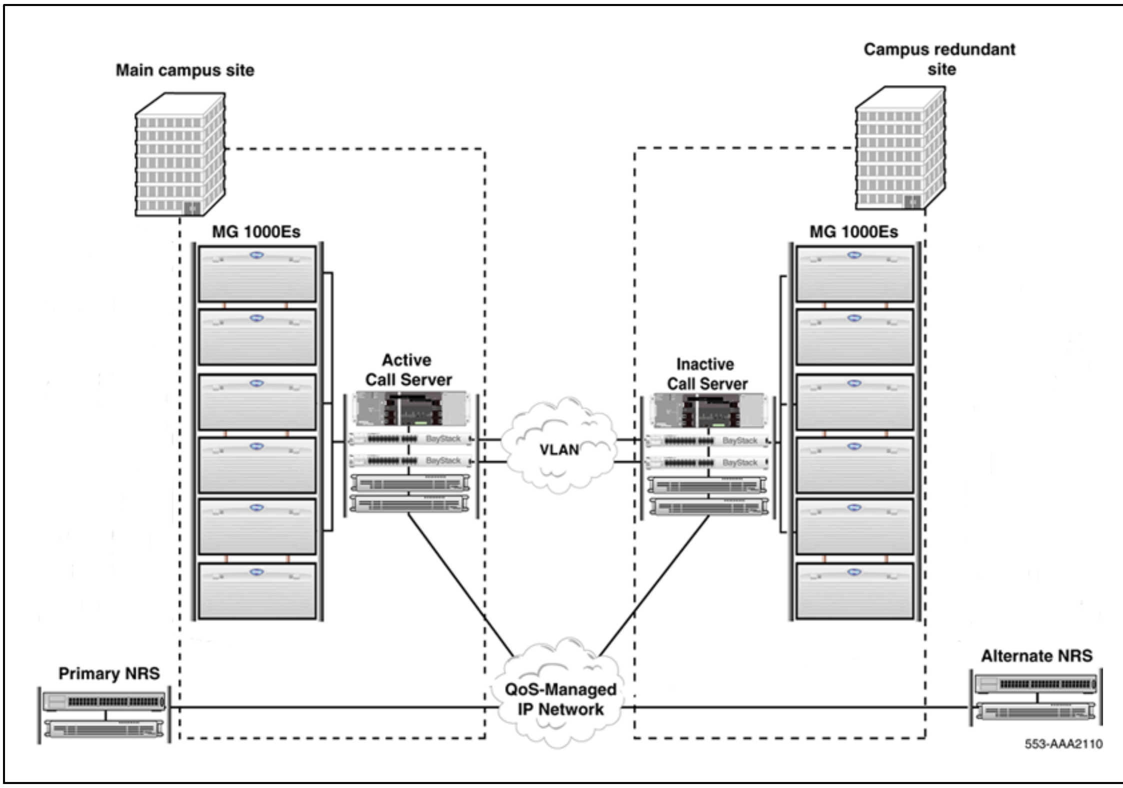
> **ATTENTION**
> Nortel has conducted extensive but not exhaustive tests of modem-to-modem calls, data transfers, and file transfers between a CS 1000E and MG 1000E, using Virtual Trunks and PRI tandem trunks. While all tests have been successful, Nortel cannot guarantee that all modem brands will operate properly over all G.711 Voice over IP (VoIP) networks. Before deploying modems, test the modem brand within the network to verify reliable operation. Contact your system supplier or your Nortel representative for more information.

Option 2: Campus Redundancy

With Campus Redundancy, customers can separate the Call Server pair across a campus IP network by extending the HSP over a network. As determined by software, the individual call processors are referred to as Call Server Core 0 and Call Server Core 1. The distance depends upon network parameter limitations specified in *System Redundancy Fundamentals (NN43001-507)*. This provides additional system redundancy within a local configuration. The Call Servers function normally and the inactive Call Server assumes control of call processing if the active Call Server fails.

To do this, the ELAN subnet and the subnet of the High Speed Pipe (HSP) are extended between the two Call Servers using a dedicated Layer 2 Virtual LAN configured to meet specified network parameters. Figure 30 "Campus Redundancy configuration" (page 84) shows a CS 1000E system in a Campus Redundancy configuration. For more information, refer to *System Redundancy Fundamentals (NN43001-507)*.

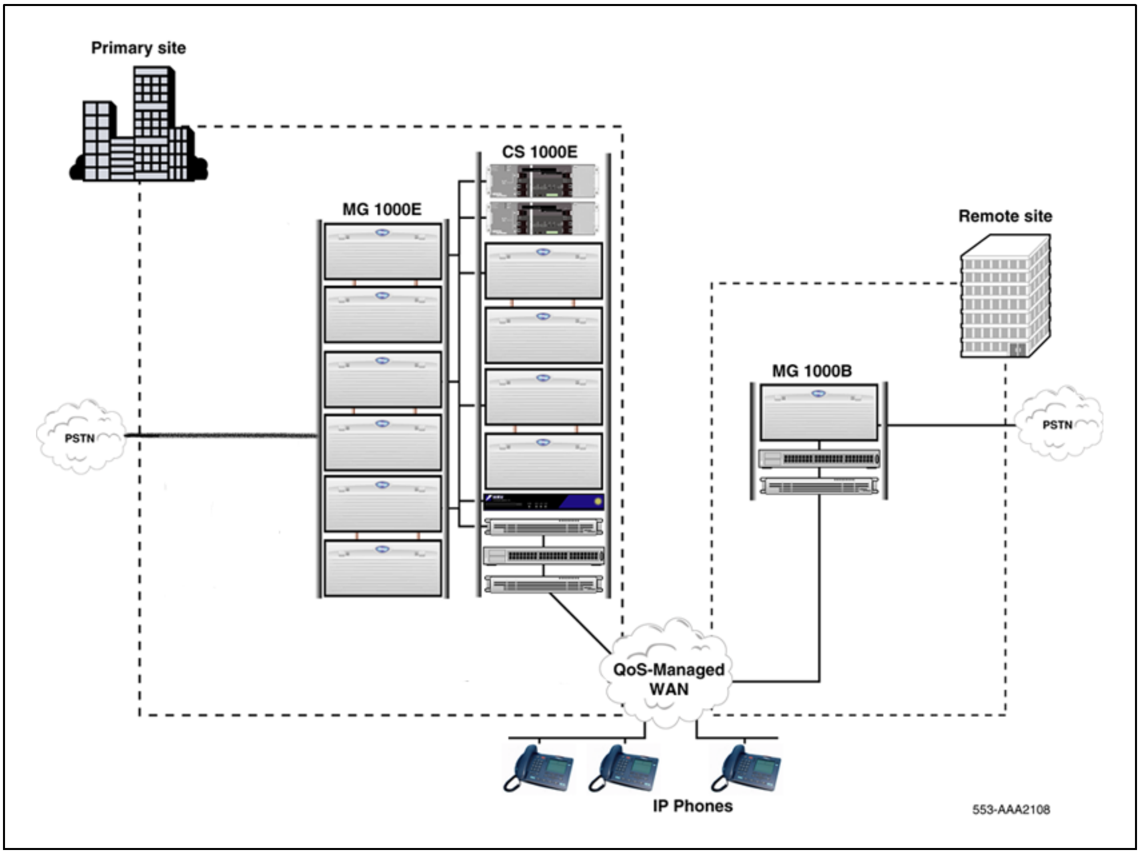
Figure 30
Campus Redundancy configuration



Option 3: Branch Office

The CS 1000E system supports the Branch Office feature, which provides central administration of Media Gateway 1000Bs (MG 1000B) at remote sites. Figure 31 "Branch Office configuration" (page 85) shows a CS 1000E system with an MG 1000B installed at a remote branch office.

Figure 31
Branch Office configuration



In this configuration, the MG 1000B is survivable. This ensures that telephone service remains available if the main office fails. For more information, refer to *Branch Office Installation and Commissioning (NN43001-314)*.

Option 4: Geographic Redundancy Survivable Media Gateway

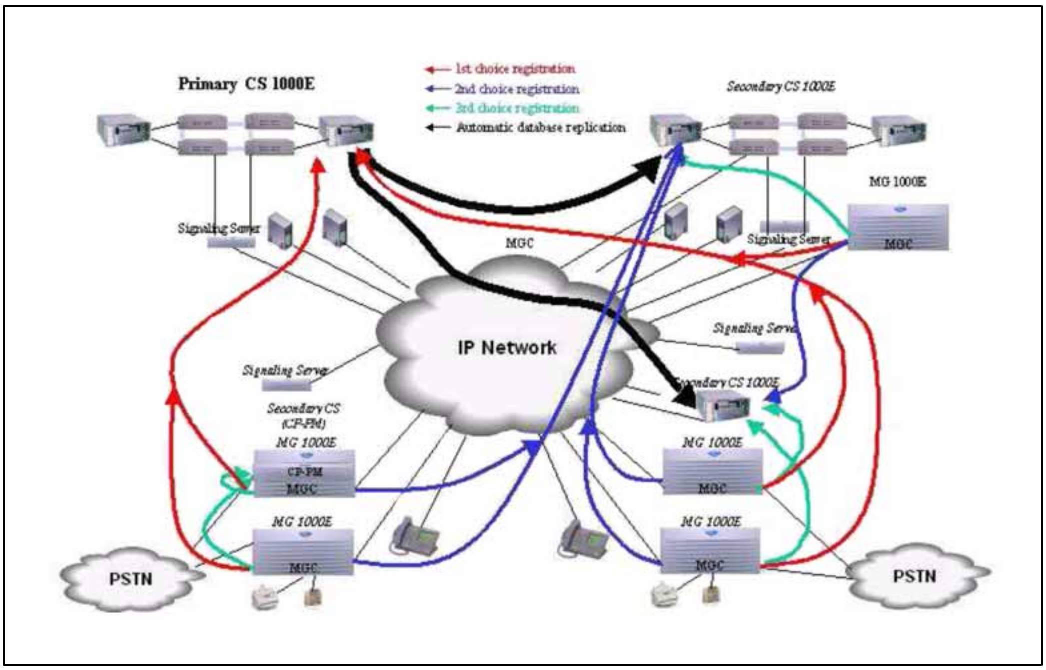
Geographic Redundancy Survivable Media Gateway is an enhancement over Geographic Redundancy. A Geographic Redundancy Survivable Media Gateway configuration consists of 1 primary Call Server and up to 50 secondary Call Servers that can be configured as alternate Call Server 1 or alternate Call Server 2.

Geographic Redundancy Survivable Media Gateway provides primary Call Server redundancy, WAN failure redundancy, and both IP and TDM resources redundancy. If the primary Call Server or WAN fails, each configured alternate Call Server provides service to the peripheral equipment and resources that it has been assigned.

The Geographic Redundancy Survivable Media Gateway replication system has a single database, administered on the primary Call Server and replicated to all secondary Call Servers. Figure 32 "Geographic Redundancy Survivable Media Gateway configuration" (page 86)shows the different paths of registration and replication.

TDM resources redundancy is achieved by "Media Gateway 1000E and Voice Gateway Media Card Triple Registration" (page 86).

Figure 32
Geographic Redundancy Survivable Media Gateway configuration



For more information about Geographic Redundancy Survivable Media Gateway, see *System Redundancy Fundamentals (NN43001-507)*.

Media Gateway 1000E and Voice Gateway Media Card Triple Registration

MG 1000Es equipped with MGC enhance redundancy by providing survivability over Layer 3 connections for network-dispersed MG 1000Es. Redundancy is performed by allowing the MG 1000E to register to either primary Call Server, alternate Call Server 1, or alternate Call Server 2.

In normal mode, all MG 1000Es in a CS 1000E are registered on the primary Call Server, and the primary Call Server provides service to all resources in the system. During a primary Call Server failure, the MG 1000E first attempts to connect to the primary Call Server without a service interruption using the Dual-Homing feature of the MGC. If a connection to the primary Call Server cannot be established, the MG 1000E reboots and registers to its configured alternate Call Server 1.

During a WAN failure, the MG 1000E reboots and registers to its configured alternate Call Server 2. Once the alternate Call Server connection is established, the MG 1000E can provide service to the resources in its own area.

When the MG 1000E is registered to any of the alternate Call Servers, it continues to poll configured Call Servers. When the primary Call Server is detected, the MG 1000E can automatically switch back to register with the primary Call Server if the registration switching policy is defined as automatic. Switching policy can also be set to manual and the MG 1000E remains registered to an alternate Call Server until a command is entered.

ATTENTION

SSC-based MG 1000Es are not compatible with Triple Registration.

With Voice Gateway Media Card Triple Registration, the Voice Gateway Media Card can register with the primary Call Server, alternate Call Server 1, or alternate Call Server 2. The Voice Gateway Media Card is configured with three IP addresses: primary, alternate 1, and alternate 2. The IP addresses of the three Call Servers must be defined on the Media Card level. To avoid the MG 1000E and Voice Gateway Media Card registering on different alternate Call Servers during a primary Call Server failure, the MG 1000E sends a message to Voice Gateway Media Card in each MG 1000E to register with the same alternate Call Server that the MG 1000E is registered with.

Planning reliability strategies

Contents

This section contains information on the following topics:

"Introduction" (page 89)

"Response to different points of failure" (page 90)

- "Call Server failure" (page 90)
- "Network failure" (page 91)
- "Signaling Server failure" (page 92)
- "NRS failure" (page 94)
- "NRS failure fail-safe" (page 95)

"MG 1000B survivability" (page 95)

- "Survival mode operation (Local Mode)" (page 96)

"CS 1000E resiliency scenarios" (page 96)

- "Call Server failure" (page 97)
- "Signaling Server failure" (page 99)
- "NRS failure" (page 101)
- "Branch Office scenarios" (page 103)

"Alternate Call Servers and survivability" (page 110)

- "IP telephony node configuration" (page 110)
- "Alternate Call Server considerations" (page 111)
- "Campus survivable MG 1000E considerations" (page 111)
- "Configuring for survivability" (page 112)

Introduction

Reliability in the CS 1000E system is based on:

1. The reliability/mean time between failures (MTBF) of components
2. Data Network robustness

3. End-point survivability
4. MGC dual-homing

Communications reliability is critical to the operation of any business. A number of capabilities are available in CS 1000E system to ensure that telephony is available when:

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

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

- Call Server with automatic database distribution by the way of configured alternate Call Servers
- IP Phone software running on a Voice Gateway Media Card in a triple registration configuration
- Signaling Server software, including Session Initiation Protocol (SIP) Gateway, H.323 Gateway, and IP Phone software
- Network Routing Service (NRS), including H.323 Gatekeeper
- Campus-distributed Media Gateways in Survival Mode

Response to different points of failure

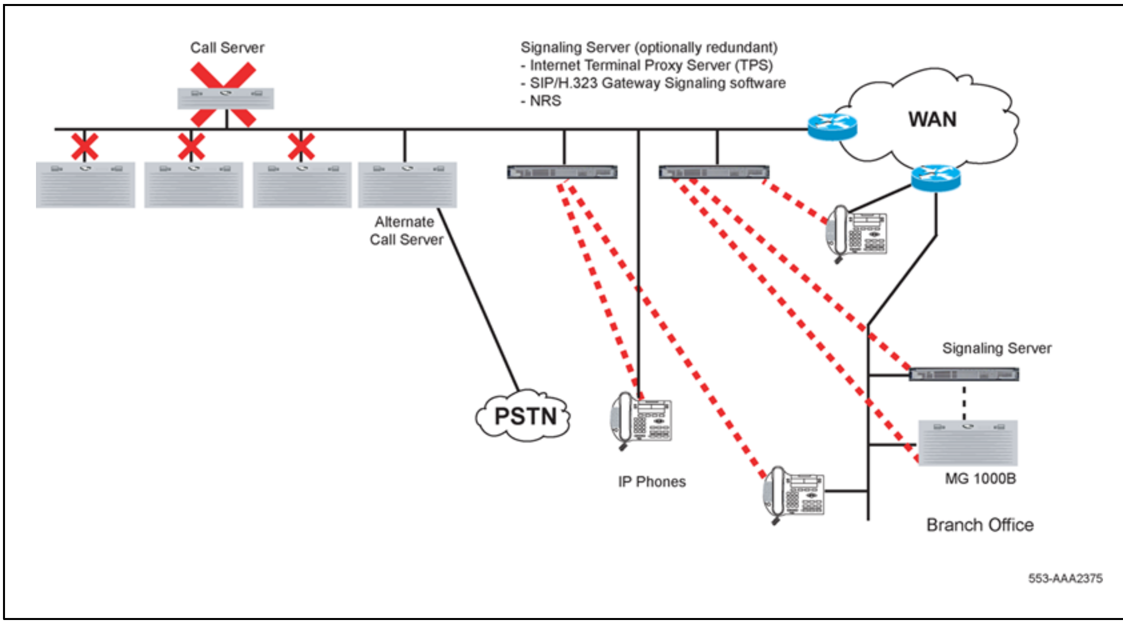
The following topics describe possible failure points and suggested remedies.

- "Call Server failure" (page 90). See "Call Server failure" (page 90).
- "Network failure" (page 91). See "Network failure" (page 91).
- "Signaling Server failure" (page 92). See "Signaling Server failure" (page 92).
- "NRS failure" (page 94). See "NRS failure" (page 94).
- "NRS failure fail-safe" (page 95). See "NRS failure fail-safe" (page 95).

Call Server failure

Figure 33 "Call Server failure" (page 91) shows a network-wide view of Call Server failure.

Figure 33
Call Server failure



Alternate Call Server

This situation applies when the CS 1000E equipment is collocated and not widely distributed.

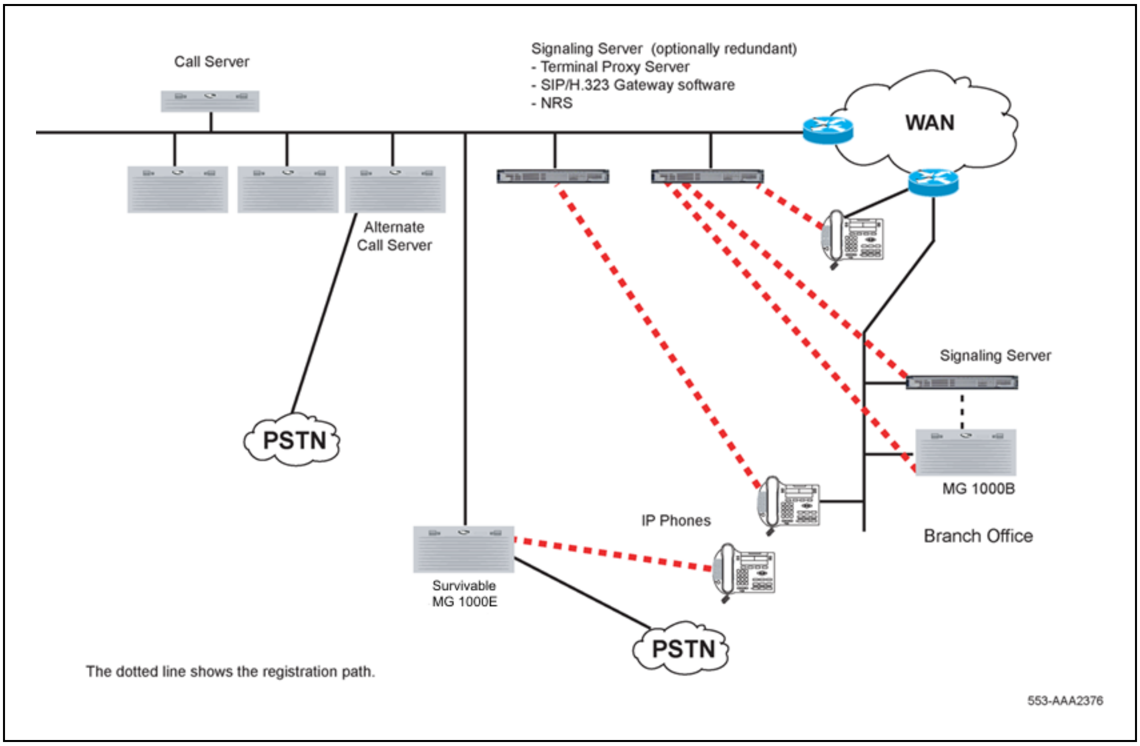
When planning reliability strategies, provision one MG 1000E as an alternate Call Server 1 within the IP Telephony node. To support an alternate Call Server 1, the installer must configure the alternate Call Server IP address in Element Manager.

If the primary Call Server fails, as shown in Figure 33 "Call Server failure" (page 91), the MG 1000E assigned as an alternate Call Server 1 assumes the role of the Call Server. The Signaling Servers register to alternate Call Server 1 and system operation resumes. Operation resumes with single MG 1000E cards, such as analog and PRI cards.

Network failure

Figure 34 "Network failure with Survivable MG 1000E" (page 92) illustrates a network failure with Survivable MG 1000E.

Figure 34
Network failure with Survivable MG 1000E



Campus-distributed MG 1000E in Survival Mode

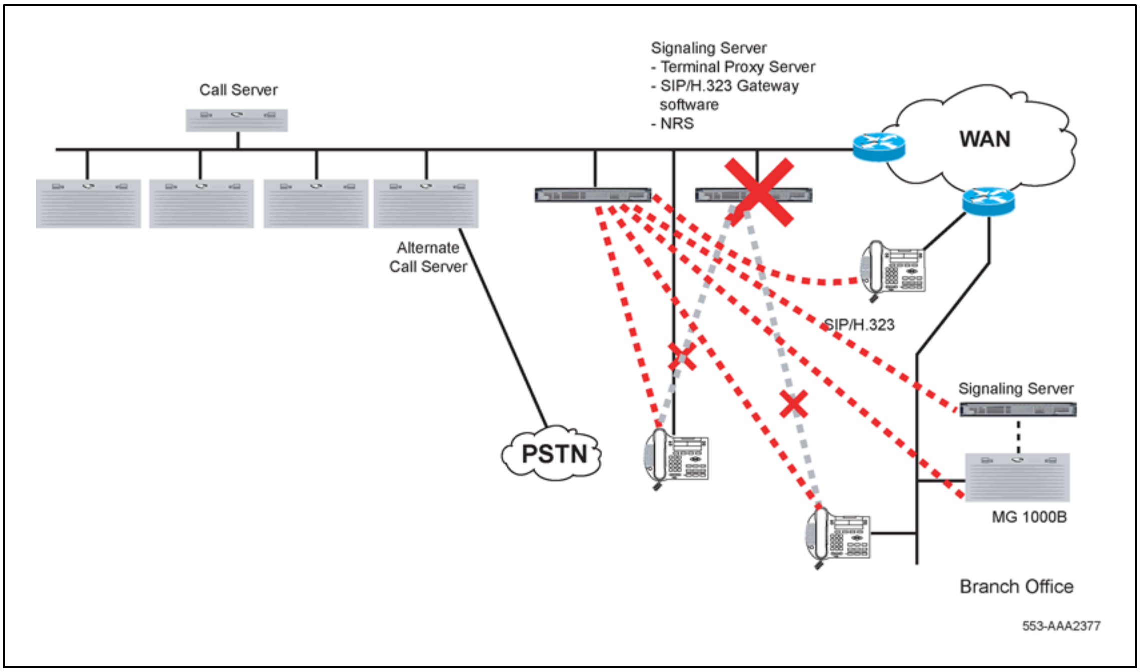
MG 1000Es equipped with CP PM can be configured as survivable when distributed throughout a campus environment. Therefore, basic telephony services can be provided in the event of a network outage. When planning for survivable MG 1000E, consider the location of critical telephones and trunks.

If the network to the IP campus fails, the remote MG 1000E shifts to the Survival Mode. The IP Phones register to the TPS on the Voice Gateway Media Cards within the remote MG 1000E. The remote IP Phones can now access the Main Office over the PSTN trunks.

Signaling Server failure

Figure 35 "Signaling Server failure" (page 93) illustrates the failure of a Signaling Server.

Figure 35
Signaling Server failure



Signaling Server redundancy

Signaling Server redundancy provides a load-sharing basis for the IP Phone Terminal Proxy Server (TPS) and an alternate route for the NRS and SIP and H.323 Gateway software.

When planning Signaling Server survivability strategies, a second or redundant Signaling Server should be installed. As shown in Figure 35 "Signaling Server failure" (page 93), two Signaling Servers can load-share when the MG 1000E contain multiple Voice Gateway Media Cards. Also, one Signaling Server is a lead Signaling Server that acts as the primary, master TPS. The other Signaling Server is a follower Signaling Server that acts as a secondary, redundant TPS, Virtual Trunk, and NRS.

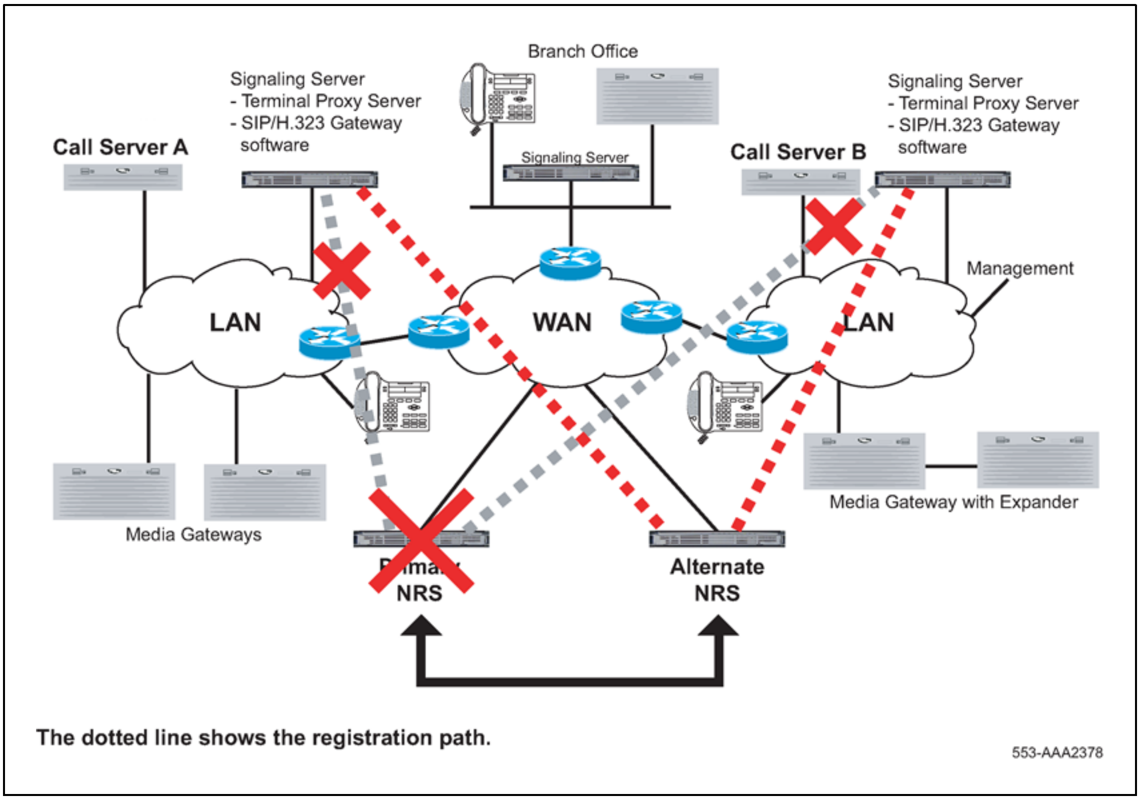
If the lead Signaling Server fails, an election process takes place and the follower Signaling Server becomes the master TPS. The IP Phones reregister to the follower Signaling Server and the system operation resumes. If the follower Signaling Server fails, the IP Phones that were registered to the follower Signaling Server reregister to the lead Signaling Server.

The same TPS functionality is available without a redundant Signaling Server. Voice Gateway Media Cards in other MG 1000E can assume a TPS role and become a source for IP Phone registration.

NRS failure

Figure 36 "NRS failure" (page 94) illustrates an NRS failure.

Figure 36
NRS failure



NRS redundancy

Figure 36 "NRS failure" (page 94) depicts a distributed environment where the TPS and NRS software reside with Call Server A and Call Server B on their own Signaling Server.

The NRS, TPS, and Gateway software can all reside on a single Signaling Server. Furthermore, primary software, the TPS, and the SIP and H.323 Gateways can all reside on Call Server A, while the second instance of NRS software can reside on a separate Signaling Server with the TPS.

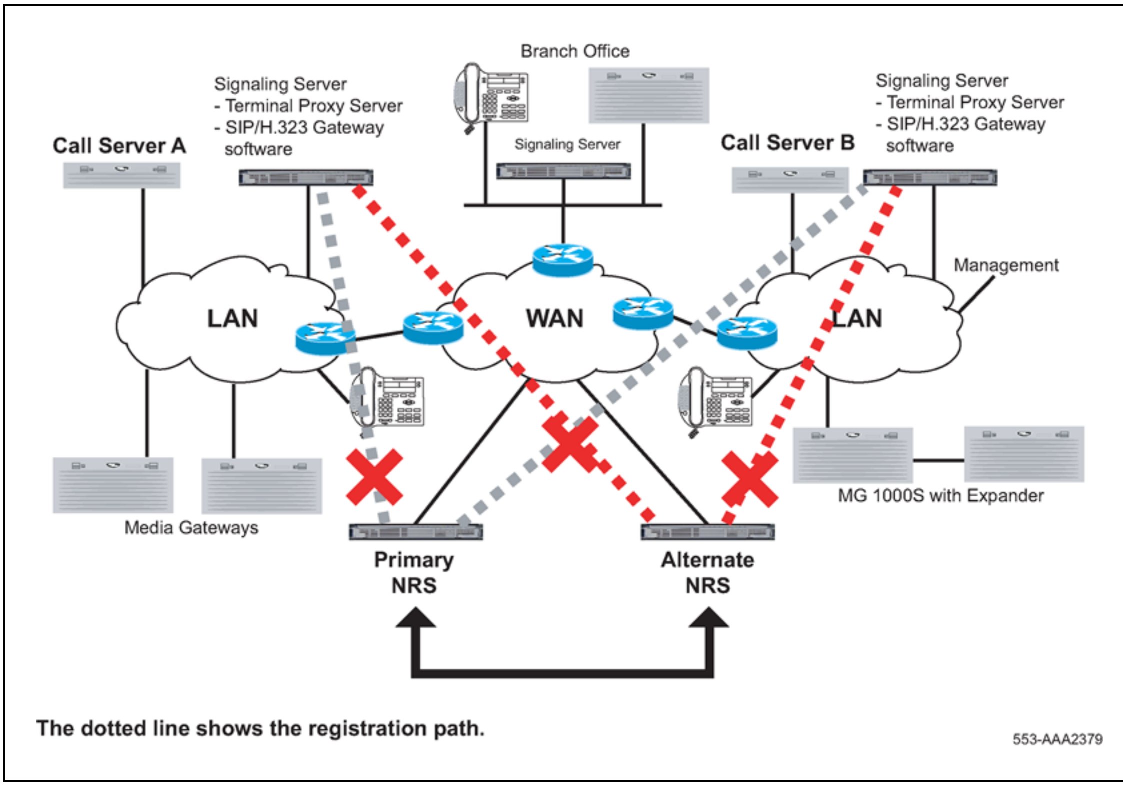
CS 1000E networks are equipped with at least one NRS to provide management of the network numbering plan for private and public numbers. An optional redundant NRS can be installed in the network. This alternate NRS automatically synchronizes its database with the primary NRS periodically.

When planning NRS survivability strategies, install a second or redundant NRS. If the primary NRS fails, the alternate NRS assumes control. The Gateways time out and register to the alternate NRS. Network calls resume.

NRS failure fail-safe

Figure 37 "NRS failure fail-safe" (page 95) illustrates NRS fail-safe.

Figure 37
NRS failure fail-safe



In addition to NRS redundancy, SIP and H.323 Gateway interfaces can withstand communication loss to both NRS by reverting to a locally cached copy of the Gateway addressing information. Since this cache is static until one NRS becomes accessible, it is only intended for a brief network outage.

The NRS can be configured as primary, alternate, or Fail-safe. If both NRS fail or a network outage to an NRS occurs, the Gateways route calls using cached data until communication to the NRS resumes.

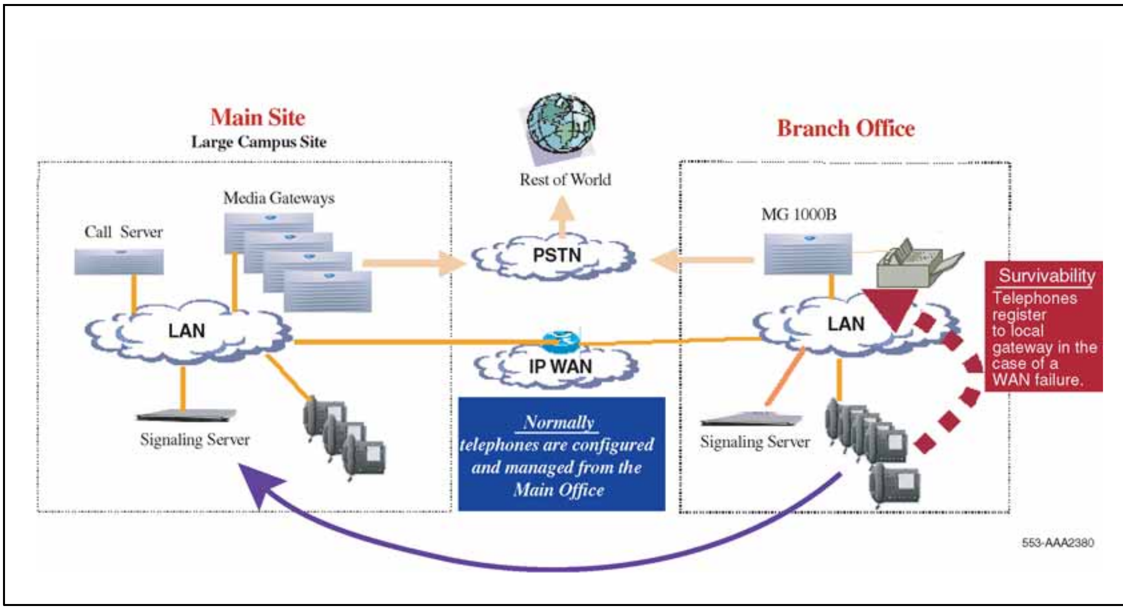
MG 1000B survivability

Refer to *Branch Office Installation and Commissioning (NN43001-314)* for details on MG 1000B survivability.

Survival mode operation (Local Mode)

If the WAN connection to the main office fails, the local MG 1000B can support call handling for the branch office IP Phones. The Media Gateway Controller (MGC) or Small System Controller (SSC) processes calls on a per-telephone basis under Local Mode (or under Test Local Mode) when system connectivity is down. In Local Mode, MG 1000B users have full access to local analog or digital trunks.

Figure 38
MG 1000B survivability



When WAN connectivity is lost, each IP Phone loses its registration with the Main Office TPS. The IP Phone reboots and registers to a TPS on the Voice Gateway Media Card in the MG 1000B.

When locally registered, the IP Phones display "Local Mode". With proper ESN configuration, MG 1000B IP Phones also access IP Phones at the Main Office through the local PSTN.

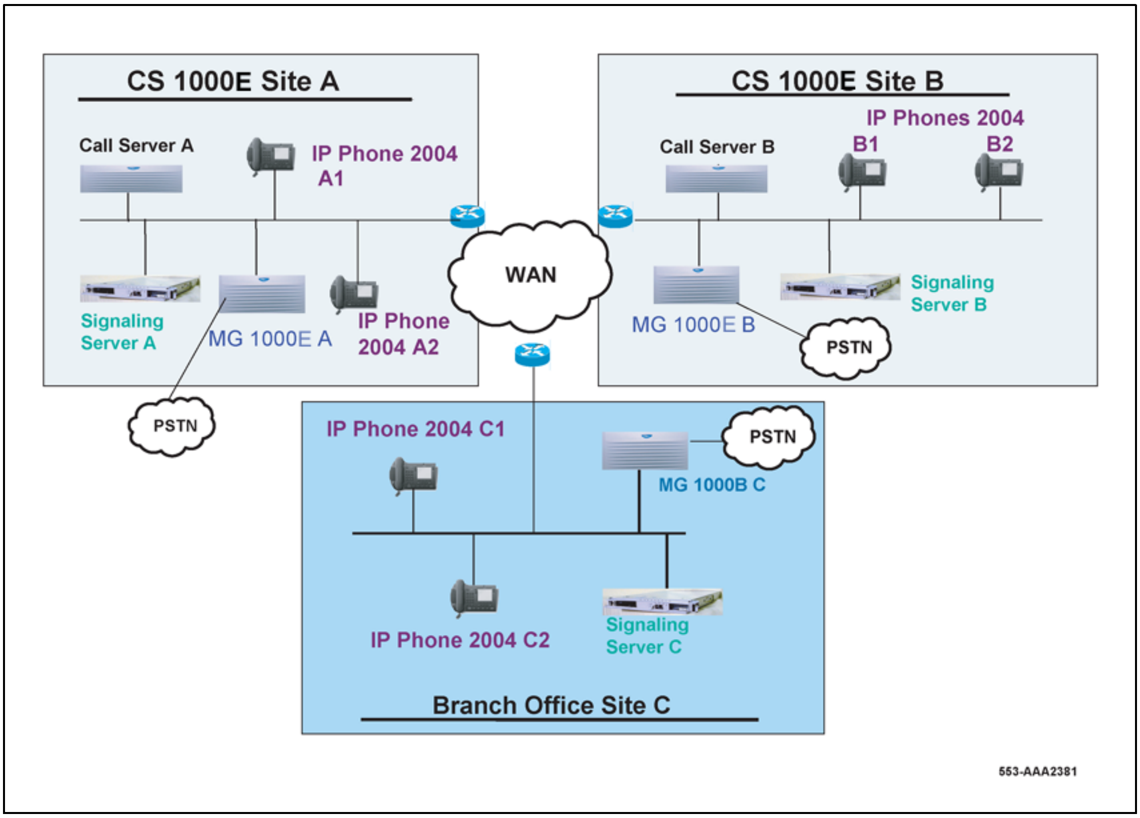
CS 1000E resiliency scenarios

This section describes the following resiliency scenarios:

- "Call Server failure" (page 97) (see "Call Server failure" (page 97))
- "Signaling Server failure" (page 99) (see "Signaling Server failure" (page 99))
- "NRS failure" (page 101) (see "NRS failure" (page 101))
- "Branch Office scenarios" (page 103) (see "Branch Office scenarios" (page 103))

Refer to Figure 39 "CS 1000E" (page 97) when reviewing these scenarios.

Figure 39
CS 1000E



Call Server failure

Resiliency Scenario 1

IP Phone 2004 A1 and A2 are talking over the LAN and Call Server A fails.

What happens to the call in progress?

The call stays up until the MG 1000E is finished rebooting, and then the call is dropped.

Describe what happens:

MG 1000E A reboots and, if it is configured as an alternate Call Server, it begins taking over all call processing. The Signaling Server reregisters to the alternate Call Server so service can be restored for all IP Phones.

Minutes before the call described in the situation can be initiated:

1.5 minutes for the MG 1000E reboot plus switchover timer. (Default for switchover timer is 2 minutes.)

Resiliency Scenario 2

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and Call Server A fails.

What happens to the call in progress?

Same as Scenario 1.

The call stays up until the MG 1000E is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 1.

MG 1000E A reboots and if it is configured as an alternate Call Server, it begins taking over all call processing. The Signaling Server reregisters to the alternate Call Server so service can be restored for all IP Phones.

Minutes before the call described in the situation can be initiated:

Same as Scenario 1.

1.5 minutes for reboot plus switchover timer.
(Default for switchover timer is 2 minutes.)

Resiliency Scenario 3

IP Phone 2004 A1 is talking to someone locally or off-net over a PSTN trunk in MG 1000E A, and Call Server A fails.

What happens to the call in progress?

Same as Scenario 1.

The call stays up until the MG 1000E is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 1.

MG 1000E A reboots and if it is configured as an alternate Call Server, it begins taking over all call processing. The Signaling Server reregisters to the alternate Call Server so service can be restored for all IP Phones.

Minutes before the call described in the situation can be initiated:

Same as Scenario 1.

1.5 minutes for reboot plus switchover timer.
(Default for switchover timer is 2 minutes.)

Signaling Server failure

Resiliency Scenario 4

IP Phone 2004 A1 and A2 are talking over the LAN and Signaling Server A fails. (Assumes no redundant Signaling Server but fail-over TPS to Voice Gateway Media Card.)

What happens to the call in progress?

The call stays up for 2.5 minutes on average and then it is dropped. Time varies due to watchdog timer on the telephone.

Describe what happens:

IP Phones type 2004 reboot and reregister with the Voice Gateway Media Card.

Minutes before the call described in the situation can be initiated:

0.5 to 1 minute after the call is dropped (assuming MG 1000E A has a Voice Gateway Media Card configured so that A1 and A2 can reregister to the Voice Gateway Media Card).

Resiliency Scenario 5

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and Signaling Server A fails. (Assumes no redundant Signaling Server but fail-over TPS to Voice Gateway Media Card.)

What happens to the call in progress?

Same as Scenario 4.

The call stays up for 2.5 minutes on average and then it is dropped. Time varies due to watchdog timer on the telephone.

Describe what happens:

Same as Scenario 4.

IP Phones type 2004 reboot and reregister with the Voice Gateway Media Card.

Minutes before the call described in the situation can be initiated:

0.5 to 1 minute after the call is dropped (assuming MG 1000E A has a Voice Gateway Media Card configured so that A1 and A2 can reregister to the Voice Gateway Media Card).

The call cannot reinitiate exactly, because in this scenario there is no longer a means of setting up a Virtual Trunk session. Therefore, the call is routed out over an alternative route (for example, PRI channel).

Resiliency Scenario 6

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and Signaling Server A fails. A redundant Signaling Server is configured on Site A.

What happens to the call in progress?

- 50% of calls on Site A stay up for 2.5 minutes, and then are dropped.
- The other 50% of telephones registered to the redundant Signaling Server on Site A do not drop the call.

Describe what happens:

- IP Phone A1 (that is, 50% of calls) reboots and then reregisters with the redundant Signaling Server.
- The other 50% have no impact on the calls in progress and the telephones stay registered to the redundant Signaling Server.

Minutes before the call described in the situation can be initiated:

- 2 to 5 minutes depending on number of telephones (2 minutes for all telephones to realize the first Signaling Server is not responding, and then all telephones from the first Signaling Server reboot and start registering with the redundant Signaling Server). At this stage, 100% of telephones from Site A are registered to the redundant Signaling Server.
- Not applicable for other 50% of telephones.

Resiliency Scenario 7

IP Phone 2004 A1 and IP Phone 2004 A2 are talking and Signaling Server A fails. A redundant Signaling Server is configured on Site A.

What happens to the call in progress?

Same as Scenario 6.

- 50% of the calls stay up for 2.5 minutes, and then are dropped.
- Other 50% of telephones registered to the redundant Signaling Server do not drop the call.

Describe what happens:

Same as Scenario 6.

- 50% of telephones on Site A1 reboot and then reregister with the redundant Signaling Server.
- Other 50% are unaffected and have no impact on the calls in progress. Telephones stay registered to the redundant Signaling Server. At this stage, 100% of the telephones from Site A are registered to the redundant Signaling Server.

Minutes before the call described in the situation can be initiated:

Same as Scenario 6.

- 2 to 5 minutes depending on number of telephones (2 minutes for all telephones to realize the first Signaling Server is not responding, and then all the telephones reboot and start registering with redundant Signaling Server).
- Not applicable for other 50% of the telephones.

NRS failure

Resiliency Scenario 8

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and the primary NRS fails. An alternate NRS is configured on Site B. Assume the primary NRS is a stand-alone box (without a TPS).

What happens to the call in progress?

The calls in progress are unaffected.

Describe what happens:

The alternate NRS takes over as Active NRS after the 30-second polling timer expires.

There is also the Time to Live timer for the H.323 endpoints to the Gatekeeper. This timer is usually configured shorter. This timer is also user configurable.

Minutes before the call described in the situation can be initiated:

New calls are established following:

- the 30-second polling timer expires
- the alternate NRS switches over to the Active NRS
- the Time to Live timer expires

Resiliency Scenario 9

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and the primary NRS (Signaling Server) fails. Assume the primary NRS is co-resident with the Signaling Server TPS on Site A. An alternate NRS is configured on Site B. Assume the alternate NRS is co-resident with Signaling Server TPS on Site B. A redundant Signaling Server is configured on Site A.

What happens to the call in progress?

Similar to Scenario 6.

- 50% of the calls on Site A stay up for 2.5 minutes, and then are dropped.
- Other 50% of the telephones registered to the redundant Signaling Server on Site A do not drop the call.
- Calls in progress are unaffected by the NRS switchover. If transient calls (for example, calls in ringing stage) exist, they are dropped due to the Signaling Server switchover.

Describe what happens:

- 50% of telephones on Site A (that is, 50% of the calls) reboot and then reregister with the redundant Signaling Server.
- Other 50% have no impact on the calls in progress and telephones stay registered to the redundant Signaling Server.
- The alternate NRS takes over as Active NRS after the 30-second polling timer expires.
- There is also the Time to Live timer for the H.323 endpoints to the Gatekeeper. This Time to Live timer is usually configured shorter than the 30-second polling timer. This timer is also user configurable. The Virtual Trunks from the first Signaling Server register to the redundant Signaling Server like the telephones.

Minutes before the call described in the situation can be reinitiated:

2 to 5 minutes depending on the number of telephones (2 minutes for all telephones to realize the first Signaling Server is not responding, and then all telephones from the first Signaling Server reboot and start registering with redundant Signaling Server). At this stage, 100% of the telephones from Site A are registered to the redundant Signaling Server.

For the other 50% of the telephones already registered to the redundant Signaling Server, new calls are established following:

- the 30-second polling timer expires
- the alternate NRS switches over to the Active NRS
- the Time to Live timer expires

Resiliency Scenario 10

IP Phone 2004 A1 and IP Phone 2004 B2 are talking and both the primary and alternate NRS fail (both are stand-alone NRS).

What happens to the call in progress?

Same as scenario 8.

The calls in progress remain unaffected.

Describe what happens:

The primary Signaling Server uses its Fail-safe NRS after it fails to register to the other NRS. At this point, the Fail-safe NRS cannot accept registrations from new endpoints.

Minutes before the call described in the situation can be initiated:

Both primary and alternate NRS timers expire. New calls are established following:

- the 30-second polling timer expires
- the alternate NRS switches over to the Active NRS
- the Time to Live timer expires

Branch Office scenarios

Resiliency Scenario 11

IP Phone 2004 C1 and C2 are talking over the LAN and Call Server A fails.

What happens to the call in progress?

The call stays up until MG 1000E A is finished rebooting, and then it is dropped.

Describe what happens:

MG 1000E A reboots at the Main Office site and acts as an alternate Call Server at Site A. The MG 1000B telephones on Signaling Server A register with the alternate Call Server.

Minutes before the call described in the situation can be initiated:

1.5 minutes for reboot plus switchover timer. (Default for timer is 2 minutes.)

Resiliency Scenario 12

IP Phone 2004 C1 and A2 are talking and Call Server A fails.

What happens to the call in progress?

Same as Scenario 11.

The call stays up until MG 1000E A is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 11.

MG 1000E A reboots at the Main Office site and acts as an alternate Call Server at Site A. The MG 1000B telephones on Signaling Server A register with the alternate Call Server.

Minutes before the call described in the situation can be initiated:

Same as Scenario 11.

1.5 minutes for reboot plus switchover timer.
(Default for timer is 2 minutes.)

Resiliency Scenario 13

IP Phone 2004 C1 and C2 are talking over the LAN and Signaling Server A fails.

What happens to the call in progress?

The call stays up for 2.5 minutes, and then it is dropped.

Describe what happens:

C1 and C2 reboot and register with the branch office Signaling Server. The telephones are redirected back to the Main Office to register with the redundant Signaling Server. If there is no second Signaling Server, the telephones register with the Voice Gateway Media Card at the Main Office site.

Minutes before the call described in the situation can be initiated:

2 to 6 minutes; 2 to 5 minutes to reboot C1 and C2, plus the extra minute for redirection.

Resiliency Scenario 14

IP Phone 2004 C1 and A2 are talking over LAN and Signaling Server A fails.

What happens to the call in progress?

The call stays up for 2.5 minutes, and then it is dropped.

Describe what happens:

A2 reboots and registers with the redundant Signaling Server at the Main Office. C1 reboots, registers with the branch office Signaling Server, and then is redirected to register with the redundant Signaling Server at the Main Office. Both A2 and C1 register with a Voice Gateway Media Card at the Main Office if there is no redundant Signaling Server. This assumes telephones are registered to the failing Signaling Server in this scenario. If 50% of telephones were registered to the surviving Signaling Server, telephones and calls would proceed as per normal healthy operation.

Minutes before the call described in the situation can be initiated:

For telephone A2, 2 to 5 minutes depending on the number of telephones (2 minutes for all telephones to realize the first Signaling Server is not responding, and then all telephones from the first Signaling Server reboot and start registering with the redundant Signaling Server or the Voice Gateway Media Card). At this stage, 100% of telephones from Site A are registered to the redundant Signaling Server or Voice Gateway Media Card.

For telephone C1, 2 to 6 minutes. The extra minute is needed to register to the branch office Signaling Server and then be redirected back to the Main Office.

Not applicable for the other 50% of telephones if registered to the redundant Signaling Server.

Resiliency Scenario 15

IP Phone 2004 C1 and C2 at the branch office are talking and the WAN data network connection to the Main Office goes down.

What happens to the call in progress?

The call stays up for 2.5 minutes, and then it is dropped.

Describe what happens:

Telephones C1 and C2 reboot and then reregister with the Signaling Server at the branch office.

Minutes before the call described in the situation can be initiated:

Minimum of 1 minute after the call is dropped. The time depends on the number of MG 1000B telephones. It is approximately 6 minutes for 400 telephones.

Resiliency Scenario 16

IP Phone 2004 C1 and A2 are talking and the WAN data network connection to the Main Office goes down.

What happens to the call in progress?

The speech path is lost as soon as the network connection is down.

Describe what happens:

A2 stays registered with Signaling Server A. C1 reboots and registers with Signaling Server at the branch office.

Minutes before the call described in the situation can be initiated:

Calls between Site A and Site C over IP only start after the WAN connection is fixed. Calls routed over PSTN Trunks can be completed as soon as the IP Phones reboot.

Resiliency Scenario 17

IP Phone 2004 C1 is talking to someone off-net over a PSTN trunk in MG 1000B C and Call Server A fails.

What happens to the call in progress?

The call stays up until MG 1000E A is finished rebooting, and then it is dropped.

Describe what happens:

MG 1000E A reboots at the Main Office site and acts as an alternate Call Server at Site A. The MG 1000B telephones on Signaling Server A register with the alternate Call Server.

Minutes before the call described in the situation can be initiated:

1.5 minutes for reboot plus switchover timer. (Default for timer is 2 minutes.)

Resiliency Scenario 18

IP Phone 2004 C1 is talking to IP Phone 2004 C2 and the branch office Signaling Server fails.

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

No impact on existing or future MG 1000B IP to IP calls in progress.

Minutes before the call described in the situation can be initiated:

Not applicable.

Resiliency Scenario 19

IP Phone 2004 C1 is talking to someone off-net over a PSTN trunk in MG 1000B C and the Signaling Server C (branch office) fails. (The behavior is the same as IP Phone 2004 A1 talking to someone off-net over a PSTN trunk in MG 1000E B and Signaling Server B fails.)

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

Telephone C1 is registered to the TPS at the Main Office site. A Virtual Trunk (SIP or H.323) session is initiated between the Signaling Server at the Main Office site and the Signaling Server at the branch office. With the loss of the Signaling Server at the branch office, the SIP or H.323 session fails. All idle Virtual Trunks become idle unregistered. Virtual Trunks that are busy on established calls also become unregistered, but they remain busy until the calls are released.

Minutes before the call described in the situation can be initiated:

If there is no redundant Signaling Server in the branch office, calls of this type cannot be initiated until the Signaling Server is reestablished. The call would, in this instance, be routed out over an alternative PSTN route.

Resiliency Scenario 20

A digital telephone in the branch office is talking to someone off-net over a PSTN trunk in MG 1000B C and the Signaling Server C (branch office) fails.

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

The call from the digital telephone proceeds as normal. The Signaling Server does not participate in this call.

Minutes before the call described in the situation can be initiated:

Not applicable.

Resiliency Scenario 21

A digital telephone in the Main Office is talking to someone off net over a PSTN trunk in MG 1000B C and Signaling Server A (Main Office) fails. A redundant Signaling Server is installed at Site A. (This is the same as a digital telephone in MG 1000E A talking to someone off-net over a PSTN trunk in MG 1000E B and Signaling Server A fails.)

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

The call from the digital telephone proceeds as normal. The Signaling Server at Site A fails, the Virtual Trunk (SIP or H.323 session) required to continue the call continues. All idle Virtual Trunks become idle unregistered and then register with the redundant Signaling Server installed at Site A. Virtual Trunks that are busy on established calls also become unregistered, but they remain busy. There is no impact on the media path between the DSP connected to digital telephone in the Main Office and that connected to the PSTN trunk. When the call is released by the user, the Virtual Trunk in the Main Office becomes idle, and then registers with the redundant Signaling Server installed at Site A.

Minutes before the call described in the situation can be initiated:

The call from the digital telephone proceeds as normal with no delay. The redundant Signaling Server at Site A initiates the Virtual Trunk (SIP or H.323 session) required to complete the call.

Resiliency Scenario 22

A digital telephone in the Main Office Site A is talking to someone off-net over a PSTN trunk in MG 1000B C and Signaling Server A (Main Office) fails. No redundant Signaling Server is installed at Site A. PSTN is configured as an alternate route. (This is the same as a digital telephone in MG 1000E A talking to someone off-net over a PSTN trunk in MG 1000E B and Signaling Server A fails.)

What happens to the call in progress?

No impact to the call in progress.

Describe what happens:

All idle Virtual Trunks become idle unregistered. Virtual Trunks that are busy on established calls also become unregistered, but they remain busy until the calls are released. There is no impact on the media path between the DSP connected to the digital telephone and that connected to the PSTN trunk.

Minutes before the call described in the situation can be initiated:

The call from the digital telephone proceeds as normal. The PSTN from the Main Office site is used as an alternative route to complete the call.

Resiliency Scenario 23

A digital telephone in the Main Office Site A is talking to a digital telephone in MG 1000B C and Signaling Server A (Main Office) fails. No redundant Signaling Server is installed at Site A. PSTN is configured as an alternate route. (This is the same as digital telephone in MG 1000E A talking to digital telephone in MG 1000E B and Signaling Server A fails.)

What happens to the call in progress?

No impact to the call in progress.

Describe what happens:

All idle Virtual Trunks become idle unregistered. Virtual Trunks that are busy on established calls also become unregistered, but they remain busy until the calls are released. There is no impact on the media path between the DSP connected to digital telephone in Main Office Site A and that connected to digital telephone in MG 1000B C.

Minutes before the call described in the situation can be initiated:

The call from the digital telephone proceeds as normal with no delay. The PSTN is used as an alternative route to complete the call.

Resiliency Scenario 24

A digital telephone in the Main Office Site A is talking to IP Phone C1 in MG 1000B Cand Signaling Server A (Main Office) fails. There is no redundant Signaling Server installed at Site A. Voice Gateway Media Cards are installed in Site A.

What happens to the call in progress?

The call stays up for 2.5 minutes on average and then it is dropped. Time varies due to watchdog timer on the IP Phone.

Describe what happens:

IP Phones type 2004 reboot and reregister with Voice Gateway Media Cards at the Main Office site by way of the branch office TPS.

Minutes before the call described in the situation can be initiated:

The call from the digital telephone proceeds as normal once the IP Phone has rebooted, 0.5 to 1 minute after the original call is dropped.

Alternate Call Servers and survivability

The CS 1000E system can be provisioned with 50 secondary Call Servers providing alternate Call Server connections if the primary Call Server becomes unavailable. An MG 1000E has two modes of operation:

- Normal Mode: the local resources of the MG 1000E are controlled by the primary Call Server call processing
- Survival Mode: the MG 1000E configured as an alternate Call Server performs call processing for its local resources

Configure IP telephony nodes and survivable MG 1000E for optimal operational efficiency and reliability.

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 MG 1000E. Therefore, several MG 1000Es can belong to the same node. Alternately, each MG 1000E can have its own node.

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

alternate Call Servers are MG 1000Es with MGC or SSC that are configured as survivable.

The survivable MG 1000E (alternate Call Server) IP address must be on the same ELAN subnet as the Call Server. If the MG 1000E 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 *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

If there are different nodes in different MG 1000E, 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, collocated) 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 1000E 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 1000E.

Refer to *IP Line Fundamentals (NN43100-500)* for a description of the enhanced redundancy for IP Line nodes feature.

Alternate Call Server considerations

The following are alternate Call Server considerations:

- MG 1000E are collocated.
- Configure one IP telephony node for the system (that is, all MG 1000E).
- Only one IP Phone Connect Server (on Voice Gateway Media Card and/or Signaling Server) is required for the node.
- Trunks in any MG 1000E can be used by all users.
- Voice gateway channels (on Voice Gateway Media Cards) in any MG 1000E can be used by all users.
- Configure one survivable MG 1000E as the alternate Call Server for the node.
- In Normal Mode, IP Phones register with the primary Call Server.
- In Normal mode, calls can be made between all MG 1000Es.
- In Survival Mode, IP Phones register with an alternate Call Server and can only use its resources.
- In Survival mode, calls cannot be made between MG 1000Es, but all their local telephones and trunks are functional.
- Less administration is required since there is only one node to manage.

Campus survivable MG 1000E considerations

The following are campus survivable MG 1000E considerations:

- MG 1000Es are in different locations.
- Configure a separate IP telephony node for each MG 1000E.
- Each MG 1000E requires an IP Phone Connect Server (on Voice Gateway Media Card and/or Signaling Server).
- At each MG 1000E, provision trunks to distribute traffic and for survivability.
- At each MG 1000E, provision voice gateway channels (on Voice Gateway Media Cards).

- Configure each survivable MG 1000E as an alternate Call Server for its node.
- In Normal Mode, IP Phones register with the primary Call Server.
- In Normal Mode, calls can be made between all MG 1000E.
- In Survival Mode, IP Phones register with an alternate Call Server and can only use its resources.
- In Survival Mode, calls cannot be made between MG 1000Es, but all their local telephones and trunks are functional.
- More administration is required since there is more than one node to manage.

Configuring for survivability

Refer to *Communication Server 1000S: Installation and Configuration (553-3031-210)* and *System Redundancy Fundamentals (NN43001-507)* for information on configuring Survivability and descriptions of its operation.

Telephony planning

Contents

This section contains information on the following topics:

"Installation planning" (page 113)

"Milestone chart" (page 114)

"Evaluating existing telephony infrastructure" (page 115)

"Telephony planning issues" (page 115)

"Numbering plans" (page 116)

"DTI/PRI clocking" (page 117)

"Clocking operation" (page 124)

"Installation and configuration" (page 128)

Installation planning

Use Table 4 "installation planning" (page 113) as a guide to prepare a detailed plan for every installation.

Table 4
installation planning

Procedure	Requirements
Research	Determine requirements for fire protection and safety, the equipment room, grounding and power, and cables.

Procedure	Requirements
Site planning	Select a site with suitable qualifications. Develop the site to meet requirements. Prepare the building cabling plan.
Delivery and installation preparation	Perform pre-installation inspections. Examine the delivery route. Review equipment handling precautions. Gather all delivery items.

Milestone chart

Site preparation activities are easier to plan and monitor when a milestone chart is used. A milestone chart is a general schedule that shows all required activities in order, with a start and end date for each. Individual operations and an overall installation schedule should both be represented. Table 5 "Milestone chart" (page 114) lists typical activities in a milestone chart. For a complex site, a more detailed chart can be required.

Table 5
Milestone chart

Step	Action
1	Select the site and complete planning activities. • Plan fire prevention and safety features. • Plan the equipment room layout. • Plan grounding and power. • Plan cable routes and terminations. • Plan and start any renovations to the equipment room.
2	Continue site construction and renovation tasks. • Install grounding, power, air conditioning, and heating. • Install special rigging, such as overhead cable racks and distribution frame equipment, as required. • Test site wiring to ensure that minimum requirements are met.
3	Complete construction and ensure that grounding and power are in place. • Test air conditioning and heating systems. • Make equipment delivery arrangements. • Complete equipment room inspection, identifying and resolving any delivery constraints.

Evaluating existing telephony infrastructure

TO determine the best way to deploy a CS 1000E system, you must evaluate the existing Telecom infrastructure. This evaluation helps decide whether to replace existing network components or add new CS 1000E system components.

The Telecom infrastructure analysis examines the products, services, and features used in the existing environment, including:

- PBX systems and locations
- system and network level features
- existing dial Plan
- supported applications
- key systems
- PBX inter-connectivity
- telephone users and features
- PSTN trunking

Telephony planning issues

To deploy the CS 1000E system, you must address the following planning issues.

- **Desktop features**. For details about desktop features, see the following:
  — *Telephones and Consoles Fundamentals (NN43001-567)*
  — *IP Phones Fundamentals (NN43001-368)*
- **System features**. For details about feature operation, see the *Feature Listing Reference (NN43001-111)*. For details about feature configuration, see the *Software Input Output Administration (NN43001-611)*.
- **System interworking and networking**. For details about Numbering/Dial Plan Configuration, see the following:
  — *Electronic Switched Network Signaling and Transmission Guidelines (NN43001-280)*
  — *Dialing Plans Reference (NN43001-283)*
  — *Basic Network Feature Fundamentals (NN43001-579)*
- **PRI/DTI clocking**. For details about PRI/DTI clocking, see the following:
  — *ISDN Primary Rate Interface Fundamentals (NN43001-569)*
  — *ISDN Basic Rate Interface Feature Fundamentals (NN43001-580)*

Applications

For details about CallPilot, Symposium, and other applications, see the following:

- *Automatic Call Distribution Fundamentals (NN43001-551)*
- CallPilot 555-7101- xxx series NTPs
- Symposium 297-2183-xxx series NTPs
- Remote Office 555-8421-xxx series NTPs
- MDECT 553-3601-xxx series NTPs
- other applications NTPs

Access

For details about signaling (ISDN-PRI, EIR2, CCS and CAS), see the following:

- *ISDN Primary Rate Interface Fundamentals (NN43001-569)*
- *ISDN Basic Rate Interface Feature Fundamentals (NN43001-580)*

For details about FXS, FXO, or ground/loop start COT trunks, see the *Circuit Card Reference (NN43001-311)*.

Numbering plans

A CS 1000E network can use many numbering plans, depending upon dialing preferences and configuration management requirements. Primary options include:

- Uniform Dialing Plan (UDP)
- Coordinated Dialing Plan (CDP)
- Transferable Directory Numbers (TNDN)

Refer to *IP Peer Networking Installation and Commissioning (NN43001-313)* for information about the following:

- the Network Routing Service (NRS) and how it performs address translation
- numbering plans
- call routing
- zoning plans
- collaborative servers

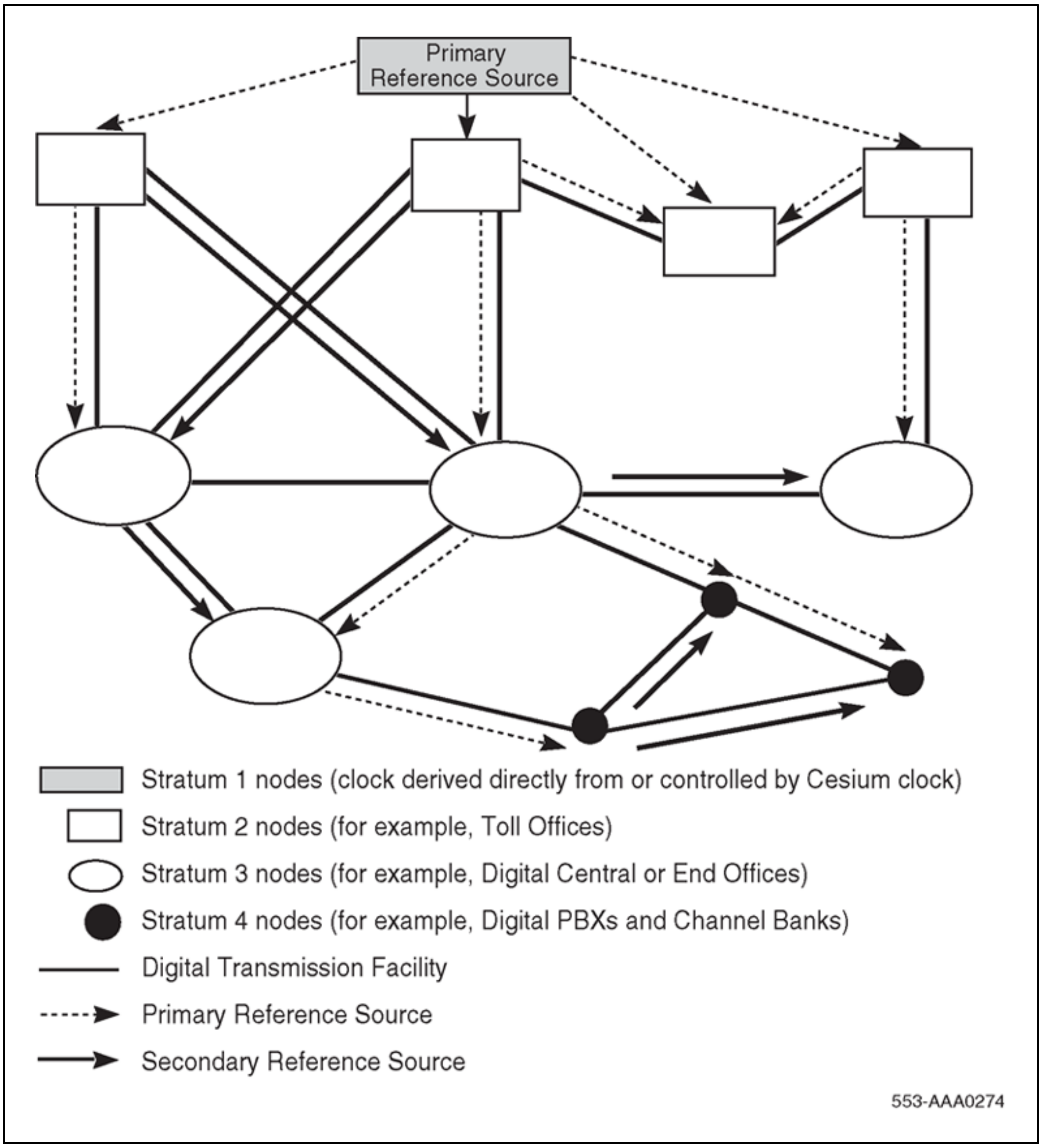
For more information about dialing plans, refer to *Dialing Plans Reference (NN43001-283)*.

DTI/PRI clocking

When digital signals transport over a digital communication link, the receiving end must operate at the same frequency as the originating end to prevent data loss, this is called link synchronization. If one end of a communication link is not synchronized, data bit slips occur and data loss results. To ensure reliable data transfer, accurate timing is important and synchronized timing is critical.

When only two PBX systems interconnect in an isolated private network, the two systems can operate in master-slave mode to achieve synchronization. In master-slave mode, one system derives its timing from the other. Slips can be lessened by forcing all systems to use a common reference clock through a network clocking hierarchy, shown in Figure 40 "Hierarchical synchronization" (page 118).

Figure 40
Hierarchical synchronization



Synchronization methods

There are two common methods of maintaining timing synchronization between switching systems:

- Pleisiochoronous operation
- Mesochronous operation

Pleisiochoronous operation

In pleisiochoronous mode, nodal clocks run independently (free-run) at the same nominal frequency. Frequency differences between clocks result in frame slips. The magnitude of frame slips is directly proportional to the difference in frequency. Slips, though inevitable, can be minimized by using stable clocks and elastic stores or buffers. The buffers absorb data bits to compensate for slight variances in clock frequencies.

Mesochronous operation

In mesochronous mode, nodal clocks are commonly and automatically locked to an external reference clock, yielding virtually slip-free operation. With this method, frame slips are eliminated if elastic stores are large enough to compensate for transmission variances.

If the CS 1000E system *is not used* as a master in a private network, Nortel recommends that systems be configured in mesochronous mode. To do this, users can configure the clock controller circuit cards to lock onto an external reference source.

If the CS 1000E system *is used* as a master in a private network, end-users can configure the system in pleisiochoronous mode. Since a private network has no digital links to a higher node category, a CS 1000E clock controller in an isolated private network can operate in free run mode and act as a master clock. Other PBX systems in the private network can then track the master clock.

North American hierarchical synchronization

Figure 40 "Hierarchical synchronization" (page 118) provides a general view of clock synchronization in a digital network, including the four Stratum levels, where Stratum 1 offers the highest accuracy and Stratum 4, the lowest. Also shown in Figure 40 "Hierarchical synchronization" (page 118) are ways to provide a secondary clock source to prevent timing loops that can cause instability in digital network synchronization.

Timing reference

In the North American network, the Primary Timing Reference is derived from a cesium beam atomic clock.

In Canada, the digital network is divided in two regions that interact plesiochronously, each with its own cesium atomic clock. Their common boundary lies between the Manitoba Telephone System and Bell Canada. The Eastern Region clock is located in Ottawa, the Western region clock in Calgary. Any DS-1 signal leaving these switches is synchronized to cesium oscillators. Every digital node in Canada (whether Central Office (CO), Digital PBX with CO connectivity, or digital Multiplexer) can trace their clock back to the cesium oscillator in Ottawa or Calgary. That is, unless the Digital System is operating in Pleisiochoronous operation.

In the United States, a similar arrangement exists. The U.S. digital network is supported by two primary clocks, one in St.Louis, Missouri, and a second in Boulder, Colorado.

Node categories/Stratum

In the North America digital network, nodes are synchronized using a priority master/slave method. Digital networks are ranked in Node Categories A to E in Canada, as shown in Table 7 "Node categories" (page 121), and in Stratum levels 1 to 5 in USA. Each node is synchronized to the highest ranking clock where the node has a direct link.

Table 6
Stratum data

	Stratum 2	Stratum 3	Stratum 4
Accuracy	+/- 1.6 * 10^-8 Hz	+/- 4.6 * 10^-6 Hz	+/- 3.2 * 10-5 Hz
Holdover	1 * 10^-10 per day	<=255 frame slips in 1st 24 hours	Not Required
Hardware Duplicatio n	Required	Required Non-duplicated hardware that meets all other Stratum 3 requirements is referred to as Stratum 3ND.	Not Required
MTIE During Rearrangement	MTIE <= 1 usec Phase Change Slope: <= 81 ns in any 1.326 msec	MTIE <= 1 usec Phase Change Slope: <= 81 ns in any 1.326 msec	Not Required Stratum 4 clock hardware that meets MTIE requirements during rearrangements is referred to as 4E.

	Stratum 2	Stratum 3	Stratum 4
Pull-in Range	3.2 * 10^-8 Hz	9.2 * 10^-8 Hz	6.4 * 10^-5 Hz
Dedicated Timing Required	Required	Required	Not Required

Table 7
Node categories

AT&T Stratum	Canadian Node Category	Operating Equipment
1 (Located in St. Louis and Boulder)	A (Located in Calgary and Ottawa)	Regional master with an associated cesium atomic clock.
2	B, C	International Gateway switch
3	D	Central Office/End Office, or digital PBX
4	E	Digital PBX or Multiplexers

Frame slip

Digital signals must have accurate clock synchronization for data to be interleaved into or extracted from the appropriate timeslot during multiplexing and de-multiplexing operations. A frame slip is defined as the repetition or deletion of the 193 bits of a DS-1 frame due to a sufficiently large discrepancy in the read and write rates at the buffer. Frame slips occur when clocks do not operate at the same exact speed.

When data bits are written into a buffer at a higher rate than the bits are read, the buffer overflows, known as a slip-frame deletion. When data bits are written into a buffer at a lower rate than the bits are read, the buffer runs dry or under-flows, known as a slip-frame repetition. Both occurrences are called a slip or a controlled slip. Frame slippage has a negative impact on data transfer, but can be controlled or avoided with proper clock synchronization.

Guidelines

Design guidelines for CS 1000E Network Synchronization are as follows:

- Where possible, the master Clock Source should always be from a Node Category/Stratum with a higher clock accuracy. When the PBX is connected to the CO, the CO is always the master and the PBX is the slave. For example, the PBX clock controller prompt PREF is set to the slot number of the DTI/PRI connected to the CO.
- Clock controllers within the system should not be in free-run unless they operate in a fully independent network where the source clock

controller acts as a master. Only one clock controller in the system can operate in free-run mode.

- When connecting two PBXs together with no CO connections, the most reliable PBX should be the master clock source.
- Avoid timing loops. A timing loop occurs when a clock uses as its reference frequency, a signal that is traceable to the output of the same clock. This produces a closed loop that leads to frequency instability.
- All Central Offices/PBX links that serve as clock references must offer a traceable path back to the same Stratum 1 clock source.
- If an MG 1000E has at least one DTI, PRI, or BRI trunk card, it must also have one clock controller installed. The clock controller tracks to the same traceable reference as the other MG 1000E.
- All slave clock controllers must set their primary reference (PREF) to the slot that they are installed. For example, a clock controller installed in slot 9 must have its PREF set to slot 9.
- The MG 1000E Expander does not support clock controllers.

Clock controller function and description

The NTAK20A-series clock controller meets Stratum 3 level requirements and the NTAK20B clock controller meets Stratum 4 requirements. The embedded clock controllers on the NTAK10 and NTAK79 cards meet Stratum level 4 requirements.

Clocking modes

The CS 1000E system supports up to one clock controller in each MG 1000E. Each clock controller can operate in one of two modes: tracking or non-tracking (free-run).

Tracking mode

In tracking mode, the DTI/PRI card supplies a clock reference to a clock controller daughterboard. Also, one DTI/PRI with clock controller is defined as the primary reference source for clock synchronization. The other (within the same MG 1000E) is defined as the secondary reference source (PREF and SREF in LD 73).

There are two stages to clock controller tracking:

1. tracking a reference
2. locked onto a reference

When tracking a reference, the clock controller uses an algorithm to match its frequency to the frequency of the incoming clock. When the frequencies are nearly matched, the clock controller locks on to the reference. The clock controller makes small adjustments to its own frequency until incoming frequencies and system frequencies correspond. If the incoming clock reference is stable, the internal clock controller tracks it, locks on to it, and matches frequencies exactly. Occasionally, environmental circumstances cause the external or internal clocks to drift. When this occurs, the internal clock controller briefly enters the racking stage. The green LED flashes momentarily until the clock controller once again locks on to the reference.

If the incoming reference is unstable, the internal clock controller is continually in the tracking stage, with green LED flashing continually. This condition does not present a problem, rather, it shows that the clock controller is continually attempting to lock on to the signal. If slips occur, a problem exists with the clock controller or the incoming line.

Monitoring references

Primary and secondary synchronization references are continuously monitored to provide auto-recovery.

Reference switchover

Switchover can occur with reference degradation or signal loss. When reference performance degrades to a point where the system clock is not able to follow the timing signal, the reference is out of specification. If the primary reference is out of specification but the secondary reference is within specification, an automatic switchover is initiated without software intervention. If both references are out of specification, the clock controller provides holdover.

Auto-recovery and chatter

If the command "track to primary" is given, the clock controller tracks to the primary reference and continuously monitors the quality of both primary and secondary references. If the primary goes out of specification, the clock controller automatically tracks to secondary if the secondary is within specification.

If both references are out of specification, the clock controller enters the Holdover mode and continuously monitors both references. An automatic switchover is initiated to the reference that recovers first. If primary recovers first, the clock controller tracks to the primary. If secondary recovers first, the clock controller tracks to secondary, and switches to primary if and when primary recovers. To prevent chatter due to repeated automatic switching between primary and secondary reference sources, a time-out mechanism of at least 10 seconds is implemented.

If the command "track to secondary" is given, the clock controller tracks to the secondary reference and continuously monitors the quality of both primary and secondary references. If secondary goes out of specification, the clock controller automatically tracks to primary, provided that primary is within specification.

Holdover and free-run

In the temporary absence of a synchronization reference signal, or when sudden changes occur on the incoming reference due to error bursts, the clock controller provides a stable holdover. The free-run mode is initiated when the clock controller has no record of the quality of the incoming reference clock If the command "free run" is given, the clock controller enters the free-run mode and remains there until a new command is received. Free-run mode automatically initiates after the clock controller has been enabled.

Free-run (non-tracking)

In free-run mode, the clock controller does not synchronize on any source. Instead, the clock controller provides its own internal clock to the system. Free-run mode can be used when the CS 1000E system acts as a master clock source for other systems in the network. If the CS 1000E system is a slave, free-run mode is not desirable. Free-run mode can take effect when primary and secondary clock sources are lost due to hardware faults. Administrators can invoke free-run mode by using software commands.

Faceplate LEDs

Table 8 "NTAK20 LED indications" (page 124) provides a description of the NTAK20 LEDs.

Table 8
NTAK20 LED indications

LED state	LED color	Definition
On	Red	NTAK20 is equipped and disabled.
On	Green	NTAK20 is equipped, enabled, and (a) locked on to a reference or (b) operating in free-run mode.
Flashing	Green	NTAK20 is equipped and attempting to lock (tracking mode) to a reference. If the LED flashes continuously over an extended period of time, check the clock controller stats in LD60. If the clock controller is tracking, this can be an acceptable state. Check for slips and related clock controller error conditions. If none exist, then this state is acceptable, and the flashing is identifying jitter on the reference.
Off		NTAK20 is not equipped.

Clocking operation

The CS 1000E system can support up to 50 active clock controllers, one for each Media Gateway with a PRI. However, an MG 1000E can support only one clock controller, and an MG 1000E Expander cannot support a clock controller.

Free-running clocks

Free-running clocks are allowed only if the CS 1000E system does not connect to a CO.

Figure 41 "Acceptable connection to an isolated private network with primary reference" (page 125) to Figure 43 "Acceptable connection with a combined CO and private network" (page 126) show acceptable connections.

Figure 41
Acceptable connection to an isolated private network with primary reference

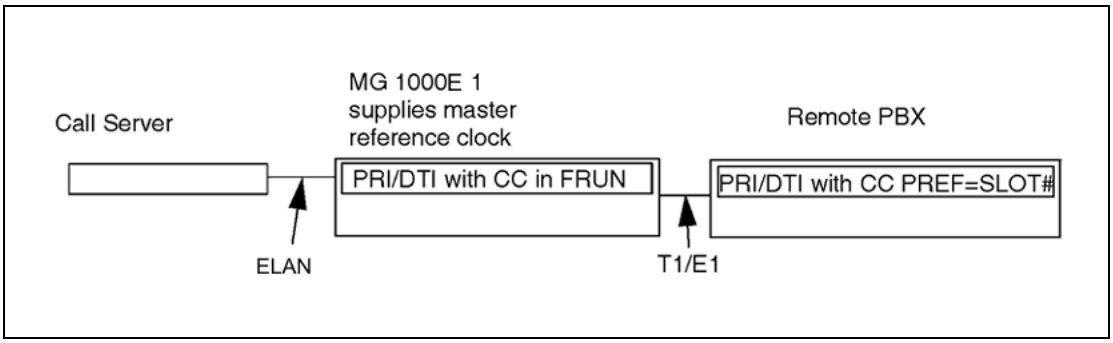


Figure 42
Acceptable connection to an isolated private system with primary and secondary reference

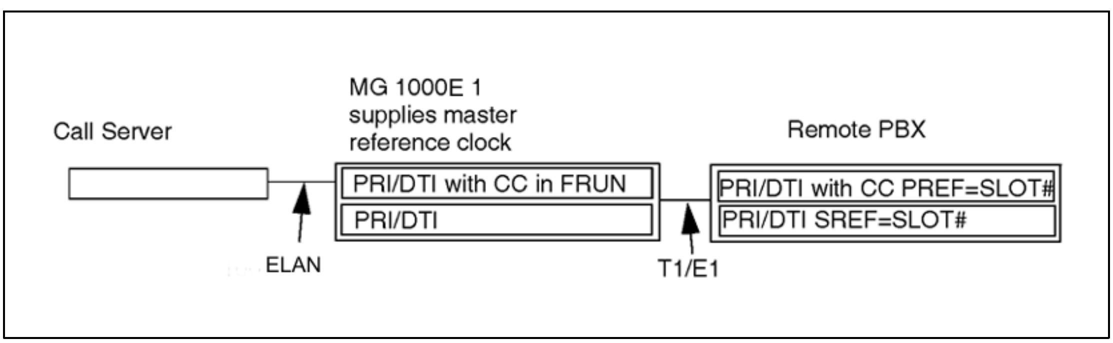
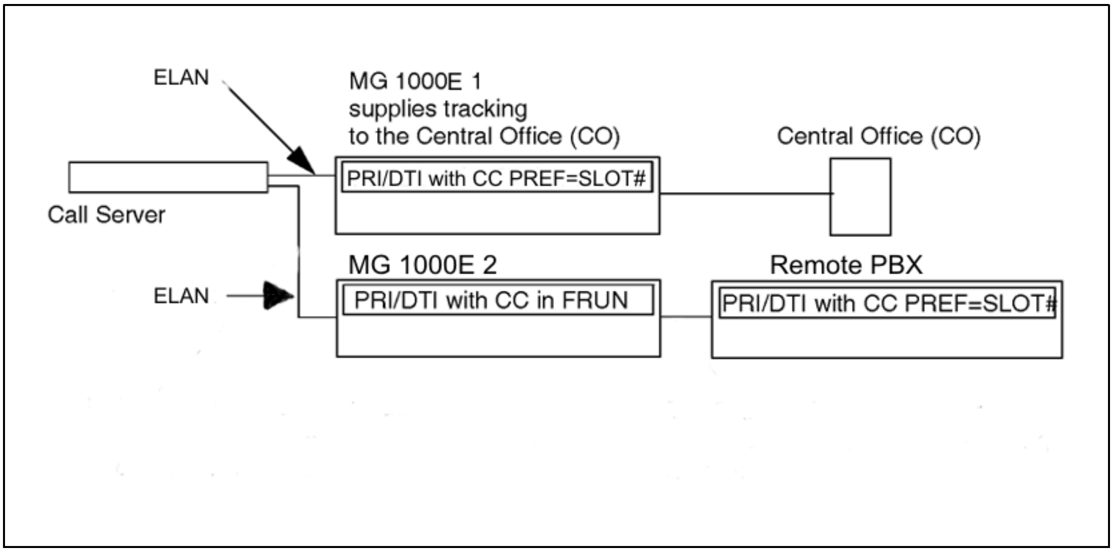


Figure 43
Acceptable connection with a combined CO and private network



Connecting to a CO

Any MG 1000E that supplies a reference to a remote PBX must have a trunk tracking to a CO. There is no clock relationship between gateways. Each media gateway operates in a separate clock domain.

Figure 44
Acceptable connection:
MG 1000E 1 and MG 1000E 2 receive clock reference directly from CO

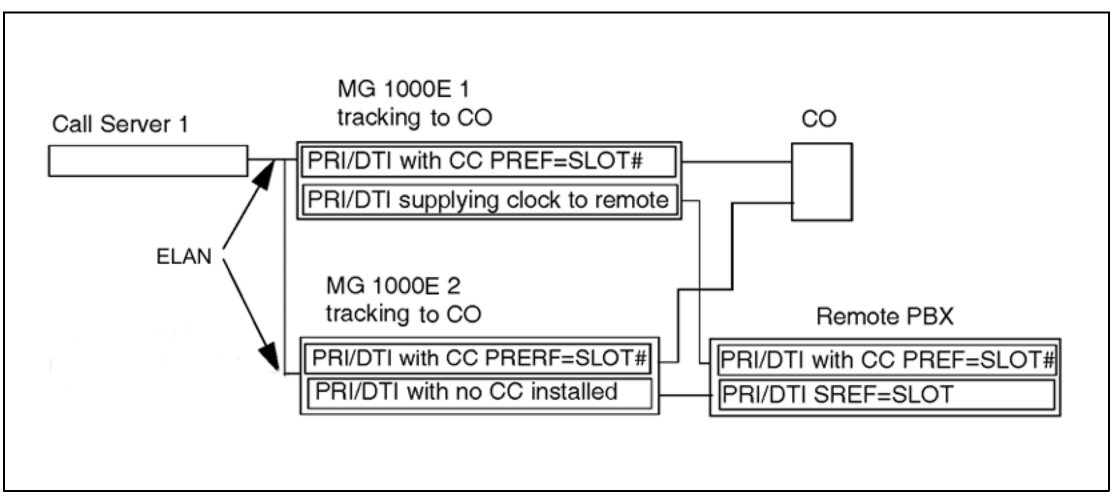


Figure 45
Acceptable connection:
MG 1000E 1 receives clock reference directly from CO/Remote and MG 1000E 2 receives clock reference indirectly from CO

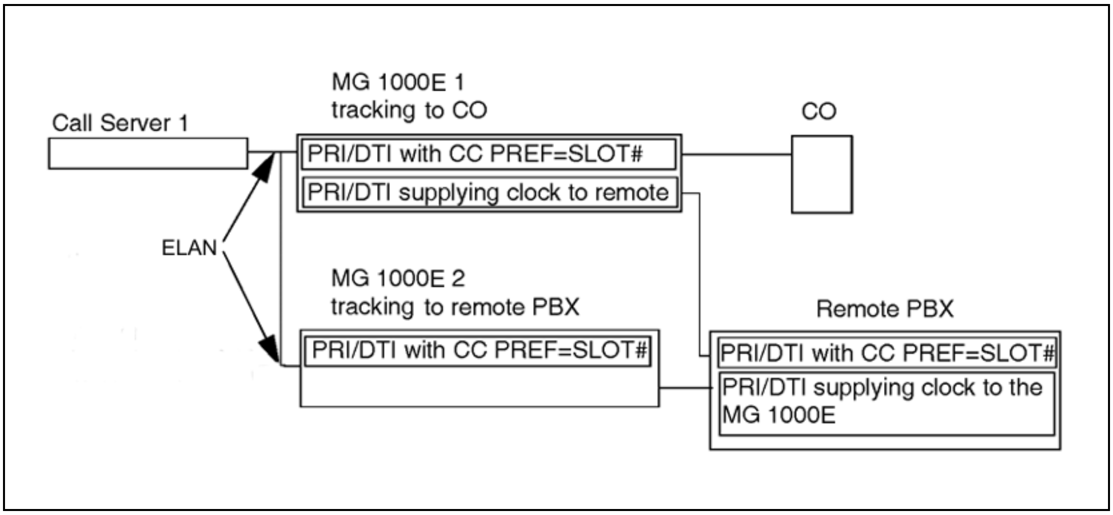
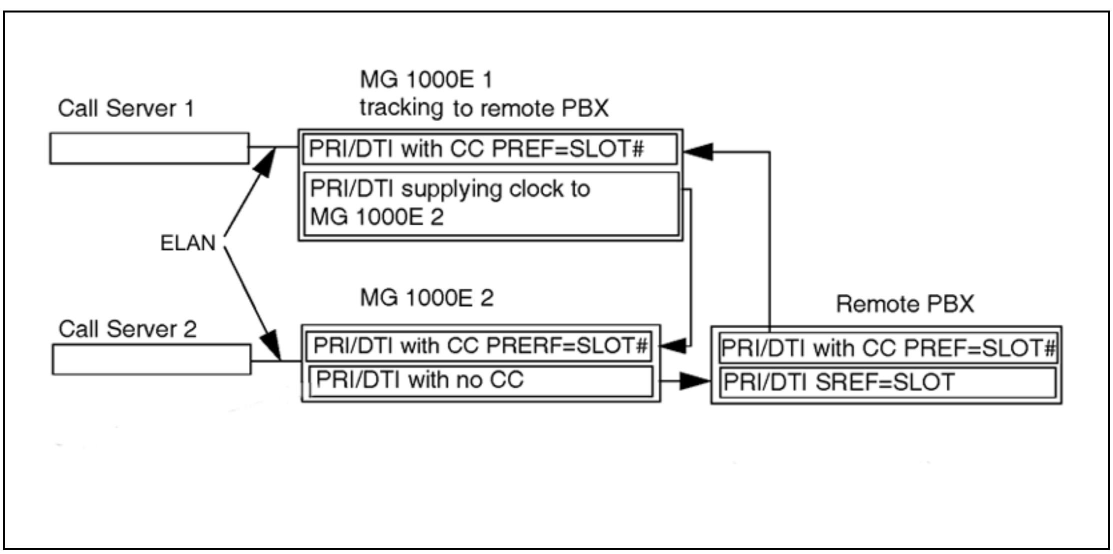


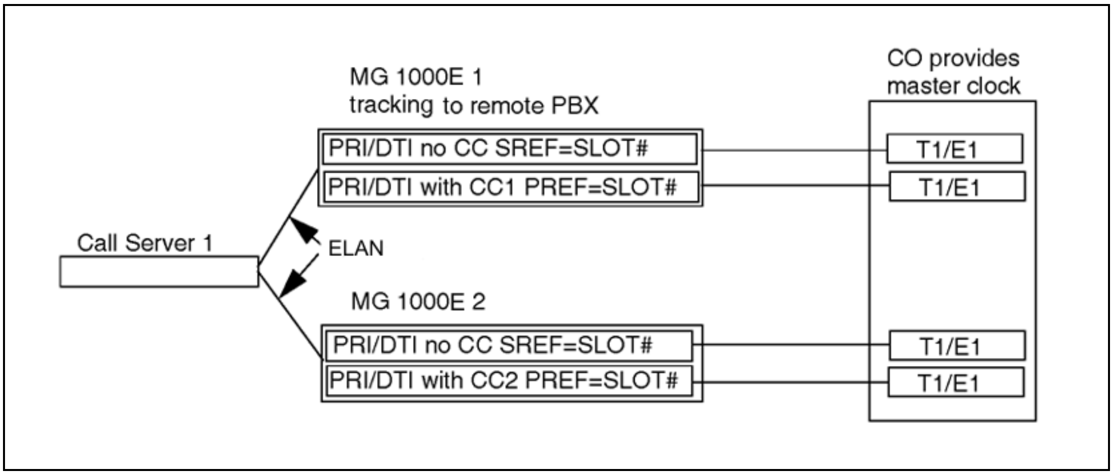
Figure 46
Unacceptable connection:
MG 1000E 1 references remote PBX; clock loop, no master clock reference



Allocating primary and secondary references

The secondary reference (SREF) clock must reside in the MG 1000E with the primary (PREF) reference.

Figure 47
Acceptable connection:
MG 1000E 1 references remote PBX; MG 1000E 2 provides master reference to remote PBX



Installation and configuration

This section describes CS 1000E system installation principles and NTAK20 clock controller daughterboard use. This section also describes installation principles and the use of 2Mb DTI/PRI embedded clock controllers.

The NTAK20 clock controller is installed on the following circuit cards:

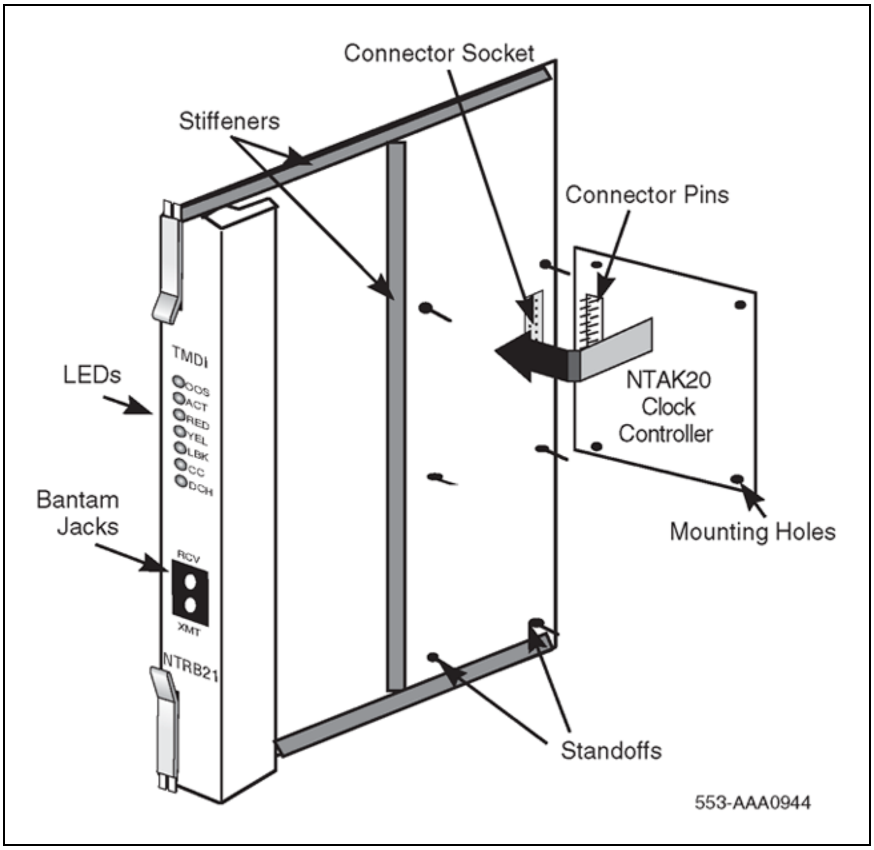
- NTRB21 – 1.5Mb DTI/PRI
- NTBK22 – BRI
- NTBK50 – 2Mb PRI

Embedded clock controllers are found on the following cards:

- NTAK10 – 2Mb DTI
- NTAK79 – 2Mb PRI

Figure 48 "NTAK20 Daughterboard installation" (page 129) shows the installation of the NTAK20 on the NTRB21 TMDI card.

Figure 48
NTAK20 Daughterboard installation



Clock controllers are configured in LD 73. For 1.5 Mb and 2 Mb DTI/PRI, the following commands are used.

LD 73 - Configure clock controllers

Prompt	Response	Description
REQ	aaa	Request (aaa = CHG, END, NEW, OUT, or PRT)
TYPE	aaaa	Type (aaaa = DTI2, PRI2, or JDMI)
FEAT	SYTI	Feature = SYTI (System Timers) Valid response when TYPE = DTI2, PRI2 or JDMI
...		
DBNC	(10)-32	Debounce timer
MGCLK	sl s c	Superloop, shelf, card number of Clock Controller

Prompt	Response	Description
PREF	card	Card of the primary reference. For MISP loop use the SILC card number. For non MISP loops use the card number entered for the MGCLK prompt.
SREF	card	Card of secondary reference. Do not use the card number entered for MGCLK or PREF prompt.

Clock Controller commands (LD 60)

Command	Description
DIS CC l s	Disable system clock controller on specified superloop and shelf.
DSCK loop	Disables the clock for loop. This is not applicable for 1.5Mb DTI/PRI.
DSYL loop	Disable yellow alarm processing for loop.
ENCK loop	Enable the clock for loop. This is not applicable for 1.5Mb DTI/PRI.
ENL CC l s	Enable system clock controller on specified superloop and shelf.
ENYL loop	Enable yellow alarm processing for loop.
SSCK l s	Get status of system clock on specified superloop and shelf.
TRCK aaa l s	Configure clock controller on IPMG specified by the superloop, loop and shelf tracking to primary, secondary or free-run. Where aaa is: • PCK = track primary clock • SCLK = track secondary clock • FRUN = free-run mode Track primary clock (PCK) or secondary clock (SCLK) as the reference clock or go to free-run (FRUN) mode.

An installed clock controller can be accessed by the following commands:

Clock Controller commands (LD 60)

Command	Description
SSCK l s	Get status of system clock on specified superloop and shelf.

Examples

Status of the CC when it is tracking to Primary.

```
.ssck 4 0
ENBL
CLOCK ACTIVE
CLOCK CONTROLLER -  LOCKED TO SLOT 1
PREF - 1
SREF -
AUTO SWREF CLK - ENBL
```

Status of the CC when it is in Free Run.

```
.ssck 12 0
ENBL
CLOCK ACTIVE
CLOCK CONTROLLER -  FREE RUN
PREF -
SREF -
AUTO SWREF CLK - ENBL
```

Status of the CC when it is tracking to Secondary.

```
.ssck 40 0
ENBL
CLOCK ACTIVE
CLOCK CONTROLLER -  LOCKED TO SLOT 2
PREF - 1
SREF - 2
AUTO SWREF CLK - ENBL
```

The tracking mode on an installed clock controller can be changed by the following commands.

Clock Controller commands (LD 60)

Command	Description
TRCK PCK l s	Configure clock controller tracking to primary on specified superloop and shelf. PCK = track primary clock Instructs the installed clock controller to track to a primary reference clock source also referred to as "SLAVE" mode.
TRCK FRUN l s	Configure clock controller tracking to free-run on specified superloop and shelf. FRUN = free-run mode

Command	Description
	Instructs the installed clock controller to free-run. In this mode, the system provides a reference or "MASTER" clock to all other systems connected through DTI/PRI links. This mode can be used only if there are no other clock controllers in SLAVE mode anywhere within the system.

The Call Server can be locked to any MG 1000E with the following command.

Clock Controller commands (LD 60)

Command	Description
TRCK PLL l s	Overrides the default search order and locks to specified superloop and shelf. Track primary clock (PCK) or secondary clock (SCLK) as the reference clock or go to free-run (FRUN) mode.

Preparing a system installation plan

Contents

This section contains information on the following topics:

"Introduction" (page 134)

"Creating an installation plan" (page 134)

"Fire, security, and safety requirements" (page 136)

"Equipment room requirements" (page 138)

"Grounding and power requirements" (page 145)

"Cable requirements" (page 145)

"LAN design" (page 147)

"Preparing a floor plan" (page 149)

"Creating a building cable plan" (page 149)

"Creating a building cable plan" (page 149)

"EC" (page 24)

"Preparing for installation" (page 154)

Introduction

> **WARNING**
> Before a CS 1000E 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 *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Planning for system installation affects the installation cost, as well as operation and maintenance, and can have an overall effect on system performance. Consider the following requirements (in addition to local and national building and electrical codes) when you plan a system installation.

Select and evaluate sites according to the requirements in this document and the following criteria:

- Space:
  - — The site must provide adequate space for unpacking, installation, operation, potential expansion, service, and storage. The site must provide space for sufficient cooling. You can need additional space for a maintenance and technician area.
- Location:
  - — The location should be convenient for equipment delivery and close to related work areas. Consider the location of related equipment, such as the distribution frame and batteries for Uninterruptable Power Supply (UPS) units. Also consider cable limitations.
- Grounding and power:
  - — Proper grounding and sufficient power facilities must be available.
- Structural integrity:
  - — The floor must be strong enough to support anticipated loads and, if applicable, the ceiling must be able to support overhead cable racks.

Creating an installation plan

To assist with the development of the installation plan, create an Installation Outline and a Milestone Chart.

Installation outline

Use Table 9 "Installation plan outline" (page 135) as a guide for preparing a detailed installation plan.

Table 9
Installation plan outline

Procedure	Requirements
Researching site requirements	• Determine fire, security, and safety requirements • Determine equipment room requirements • Determine grounding and power requirements • Determine cable requirements
Planning the site	• Prepare a floor plan • Estimate floor loading • Prepare the building cabling plan
Preparing for delivery and installation	• Prepare for delivery • Prepare for installation

Milestone chart

Planning and monitoring site preparation activities is easier when you use a milestone chart. A milestone chart is a general site planning schedule showing the sequence of activities necessary to complete a job.

Table 10 "Milestone chart" (page 135) lists typical activities included in a milestone chart. For a complex site, you must create a more detailed chart.

Table 10
Milestone chart

Task	Action
1	Select the site.
2	Plan fire prevention and safety features.
3	Plan the equipment room layout.
4	Plan grounding and power.
5	Plan cable routes and terminations.
6	Plan and start any renovations to the equipment room.
7	Continue site construction and renovation tasks.
8	Install grounding, power, air conditioning, and heating.

Task	Action
9	Install special rigging, such as overhead cable racks and distribution frame equipment, as required.
10	Test site wiring to ensure that minimum requirements are met.
11	Complete construction and ensure that grounding and power are in place.
12	Test air conditioning and heating systems.
13	Make equipment delivery arrangements.
14	Complete equipment room inspection, identifying and resolving any delivery constraints.

When you prepare a milestone chart, consider not only individual operations, but the overall installation schedule. The milestone chart should show the necessary operations in order and can assign a start and end date for each activity.

Fire, security, and safety requirements

Building, fire, and safety codes establish the degree of protection required for an installation. Additional information is available from the National Fire Protection Association (NFPA) in "Standard for the Protection of Electronic Computer/Data Processing Equipment" (NFPA 75) and "National Electrical Code (NEC)" (NFPA 70).

Fire protection and prevention

Expertise is required to properly locate and install:

- Sprinkler heads
- Fire and smoke sensing devices
- Other fire extinguishing equipment

During the planning stage, consult local codes, experts, insurance underwriters, and local building authorities.

You can implement some fire precautions when an equipment area is constructed. For example, extend walls from floor to ceiling, and construct walls, floor, and dropped ceiling of noncombustible material.

If the structural floor is made from combustible materials, cover it with a noncombustible covering and remove all debris between the raised and permanent floors before the system is installed. If there are power connections beneath a raised floor, use waterproof electrical receptacles and connectors.

You can install shatterproof windows and sprinklers outside and above the windows to keep fire from spreading from an adjacent room or building. The roof or floor above the equipment area must be watertight. Design ducts and plumbing for air-conditioning systems to keep fire, heat, and smoke from spreading from one part of a building to another. Install smoke detectors in all appropriate places.

Regularly check services such as steam, water, and power, and inspect pipes for excess condensation, leaks, or corrosion.

Fire extinguishing systems

In most cases, carbon dioxide or water sprinkler systems are the recommended fire extinguishing systems.

Dry-pipe water sprinklers are strongly recommended. This type of system interrupts power to the room and opens a master valve that fills the overhead sprinklers.

Carbon dioxide systems are also effective in containing a fire, but they quickly exhaust the oxygen supply. If you use a carbon dioxide system, you must install an alarm to warn site personnel when carbon dioxide is released. For health and safety reasons, employees must be evacuated within 30 seconds of the release.

DANGER
Nortel does not recommend using Halon or any other fire extinguishing system that is not described above.

Security precautions

You may need to extend and improve existing building security to provide adequate protection for the equipment. For example, you can install safeguards such as tamper proof keylock door controls and electrically taped glass doors and windows that can tie into an alarm system. You can also install a monitoring unit using closed-circuit television.

ATTENTION
Electric locks, such as push button access code or card reader locks, are not recommended unless you provide a battery backup or a key override.

Protect critical data, such as business records, by storing backups well away from the equipment room. A regular updating program is highly recommended.

Safety procedures and training

Company personnel should be taught how to respond to emergencies; some companies designate trained individuals as security members. Training can include when and how to evacuate personnel and records, notify the fire department, shut off all electrical power, and handle fire extinguishers properly.

In addition, install temperature and humidity monitoring devices (both visual and audible alarm signals) in equipment and storage rooms so people can respond quickly to an emergency.

Occupational noise exposure

If employees are subjected to noise levels exceeding local standards, or the levels listed in 1910.5 of the Occupational Safety and Health Administration (OSHA) Standards, initiate administrative and engineering controls. If these controls do not reduce sound levels effectively, provide protective equipment.

ATTENTION

The acoustic noise generated by a system ranges from 45 dBA to 60 dBA (decibels "A"-weighted).

Equipment room requirements

The environment for the system (and for storing spare parts) can influence system performance and reliability. Temperature, humidity, and other environmental factors, such as static electricity, must be controlled to meet system operating requirements.

Environmental requirements

The environment that the CS 1000E system operates in must meet the following general conditions:

- The room must be clean, relatively dust-free, and well ventilated. On equipment, ventilating openings must be free of obstructions.
- A Call Server can dissipate up to 40 watts of power.
- Each Signaling Server (rack mount) can dissipate up to 125 Watts of power.
- Each MG 1000E and MG 1000E expander can dissipate up to 370 Watts of power. There must be enough ventilation in the equipment room to maintain an acceptable temperature.
- For an installed Call Server, MG 1000E, MG 1000E Expander, and Signaling Server, maintain temperature between 0° and 45° C (32° and 114° F).
- Maintain humidity between 5% and 95% non-condensing.

- Select a location for equipment installation that is not subject to constant vibration.
- Locate equipment at least 12ft (3660 mm) away from sources of electrostatic, electromagnetic, or radio frequency interference. These sources can include:
  - — power tools
  - — appliances (such as vacuum cleaners)
  - — office business machines (such as copying machines)
  - — elevators
  - — air conditioners and large fans
  - — radio and TV transmitters
  - — high-frequency security devices
  - — all electric motors
  - — electrical transformers

Space requirements

Space and equipment layout requirements differ with each installation. When you plan the site, consider the following requirements:

- Primary storage
- Secondary storage
- Maintenance and technician space

Primary storage

The floor area required for a system depends on the number of racks, the length-to-width ratio of the area, and the location of walls, partitions, windows, and doors. To determine the exact layout required, prepare a detailed floor plan after regarding all of the requirements in this chapter.

Although operating needs determine the general location of terminal devices, these devices must not be located beyond the maximum distances defined by the Terminal Server. Wall jacks and outlets must be provided for all devices located in the equipment room.

Secondary storage

Provide space in the equipment area for storing disks, printer paper, printouts, and daily reports. A secure storage room for spare parts is recommended.

Whenever possible, maintain the same environmental conditions in the equipment room and storage areas. If it is not possible to maintain the environment of the storage area exactly the same as the environment of the operating equipment, give stored materials time to adjust to the equipment room environment before using them.

Maintenance and technician space

You can use the maintenance and technician area as an online work center and a place to store tools, test equipment, system documents, and spare parts. The area should have good lighting and convenient access to the system.

Typical items in a maintenance and technician area include:

- Shelves for instruction books
- Spare parts storage room
- Paper storage area
- Locking cabinet or storage area for backup disks
- Table or desk
- Terminal, printer, or equivalent device

During regular system operation, a terminal, or a modem, or both must be connected permanently to the system to provide a constant I/O interface. You can use more than one terminal or modem. Plan for surface apace, power outlets, and the availability of the terminals/modems before installation.

Temperature and humidity control

Frequent and extended system operation above recommended temperature limits can degrade system reliability. Low humidity can increase static electricity build-up, while high humidity can affect the performance of disks and printers.

Take temperature readings 76 cm (30 in.) from the front of the system. Table 11 "Operating environment" (page 141) shows system operating requirements.

> **DANGER**
> **Damage to Equipment**
> Do not expose equipment to absolute temperature limits for more than 72 hours. Do not place heat sources (such as floor heaters) near the equipment.

Table 11
Operating environment

Equipment	Temperature and humidity considerations
CS 1000E	Recommended: • 15° to 30°C (59° to 86°F) • RH 20% to 55%, non-condensing Absolute: • 10° to 45° C (50° to 113°F) • RH 20% to 80%, non-condensing • temperature change less than 10°C (18°F) per hour
Telephones	Absolute: • 0× to 50°C (32° to 122°F) • RH 20% to 80%, non-condensing
Other terminal devices (such as personal computers, data sets, and printers)	Refer to the specific documentation or manufacturer's guidelines

If you operate the system within recommended temperature limits, there are no thermal restrictions on any equipment.

Follow the specifications listed in Table 12 "Storage environment" (page 142) to store or transport equipment.

Table 12
Storage environment

Equipment	Temperature/humidity considerations
CS 1000E Call Server	• –50° to 70°C (–58° to 158°F) • RH 5% to 95%, non-condensing
Telephones	• –50° to 70°C (–58° to 158°F) • RH 5% to 95%, non-condensing
RMD (Compact Flash)	• -25° to -70°C (-13° to -94°F) • RH 8% to 95%, non-condensing
Media Gateways	• –50° to 70°C (–58° to 158°F) • RH 5% to 95%, non-condensing
Other terminal devices	Refer to the specific Nortel publication or the manufacturer's guidelines

ATTENTION
Temperature changes must be less than 30° C (54° F) per hour for storage and during transportation.

Air conditioning guidelines

Use the following guidelines to estimate air conditioning requirements. Exact requirements must be determined by a qualified air conditioning engineer.

- The air conditioning system in equipment areas must handle:
  - the heat produced by the equipment, room personnel, and lighting; and,
  - the heat that comes through walls, windows, floors, and ceilings.
- A stable ambient operating temperature of approximately 22° C (72° F) is recommended. The temperature differential in the equipment room must not exceed ±3.0° C (±5° F).

  For systems with reserve power equipment, consult the manufacturer's specifications for recommended operating temperatures.
- Heat dissipation from a system is estimated in Btu per hour (Btu/hr). You can estimate the amount of air conditioning required at a rate of one ton of refrigeration for every 12 000 Btu/hr of heat generated in the equipment area plus one ton for each 500 sq ft of floor space.

  Each person in the equipment room generates 600 Btu/hr.

> **CAUTION**
> **Damage to Equipment**
> Because digital systems require constant power (even if the system is idle), they generate heat continuously. Air conditioning requirements must be met at all times.

- Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) and Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) show the thermal dissipation for system components.

Other environmental factors

In addition to temperature and humidity, many environmental factors must be controlled in equipment areas. The environmental factors that must be controlled include:

- Static electricity
- Vibration
- Electromagnetic and radio frequency interference (EMI/RFI)
- Dust
- Lighting
- Earthquake bracing
- Structural features

Static electricity

Electronic circuits are extremely sensitive to static discharge. Static discharge can damage circuitry permanently, interrupt system operation, and cause lost data.

Static electricity can be caused by physical vibration, friction, and the separation of materials. Other common causes of static electricity build-up are low humidity, certain types of carpeting, the wax on equipment room floors, and plastic-soled shoes. The human body is the most common collector of static electricity. A combination of plastic-soled shoes, certain flooring materials, and low humidity can cause body charges in excess of 15 kV.

> **ATTENTION**
> IEEE Standard 142-1982 recommends that flooring resistance be more than 25 000 ohms and less than 1 million megohms, measured by two electrodes 0.91 m (3 ft) apart on the floor. Each electrode must weigh 2.2 kg (5 lb) and have a dry flat contact area of 6.35 cm (2.5 in.) in diameter.

Antistatic wrist straps, sprays, and mats are available. Nortel recommends at least using an antistatic wrist strap whenever you work on equipment.

Vibration

Vibration can cause the slow deterioration of mechanical parts and, if severe, can cause serious disk errors. Avoid structure-borne vibration and consequent noise transferred to the equipment room. Raised floors must have extra support jacks at strategic places to prevent the transmission of vibration.

Limit vibration in an office environment to a frequency range of 0.5–200 Hz and a G-force magnitude of 0.1 G (in accordance with the Bellcore "Network Equipment Building Systems Generic Equipment Requirements" specification TR-EOP-000063).

Electromagnetic and radio frequency interference

Sources of electromagnetic and EMI/RFI located close to equipment can cause problems with system operation. Common EMI/RFI sources known to disturb system operation include:

- Thunderstorms, static electricity, and high-voltage power lines
- Radar, broadcast stations, and mobile communications
- Power tools, appliances (such as vacuum cleaners), and office business machines (such as copiers)
- Industrial machines and ultrasonic cleaners
- Vehicle ignition, arc welders, and dielectric heaters
- Dimmer switches

Dust

Accumulated dust and dirt can degrade system reliability and performance. Dust and dirt can:

- Scratch the contacts on circuit cards causing intermittent failures
- Have conductive contents that increase static electricity in the environment
- Cause components to operate at higher temperatures

Average dust density for an office environment must be 0.00014 g/m3 or better. False ceilings and tiled floors help maintain dust density requirements.

Lighting

Lighting illumination of 50 to 75 footcandles measured 76 cm (30 in.) above the equipment room floor is recommended. Avoid direct sunlight in the equipment room to prevent malfunctions by devices with light sensors (such as disk units).

Lighting must not be powered from the equipment room service panel. For large system installations, consider provisions for emergency lighting in the equipment room.

Earthquake bracing

Earthquake (seismic) bracing is required or should be considered in some locations.

Structural features

Use sealed concrete, vinyl, or mastic tile for flooring and ensure that it meets the floor loading requirements described later in this document. Avoid using sprayed ceilings or walls.

Grounding and power requirements

Refer to "Power and grounding" (page 157) for detailed information.

Reserve power equipment room

If the reserve power equipment is located in a separate room then that room must meet the following conditions.

1. Well-ventilated and operating at optimum temperature; specific gravity readings are based on 25 degrees C (77 degrees F)
2. Located within the recommended proximity to the system
3. Equipped with protective equipment (such as goggles, face shields, acid-resistant gloves, protective aprons, water for rinsing eyes/skin, and bicarbonate of soda)
4. Well-secured
5. Accessible (the doorway must not be blocked)
6. Meet all floor loading requirements and the noise levels required by OSHA standards 1910.5 (or local standards)

For detailed instructions on battery usage, see ANSI/IEEE Standard 450-1987: "Maintenance, Testing and Replacement of Large Storage Batteries."

Cable requirements

This section describes the types of cable used in the system. It also provides some cabling guidelines.

Cable types

The system uses the following major types of wiring:

- 25-pair main distribution frame (MDF) cables:
  These cables carry voice and data information between gateways and the distribution frame. One end of the cable must be equipped with a 25-pair female connector that terminates on the module input/output (I/O) panel. The other end of the cable terminates on the MDF block.
- Interface cables:
  Interface, or I/O, cables are typically 25-conductor interfaced through RS-232-C connectors. These cables are used to connect data units to printers, host computers, and modems.
- Three port cables:
  This cable is used as an interconnect between terminal equipment and the terminal port on the Media Gateway. The cable also functions as a remote TTY if it has been configured with an MGC. On the MG1000E, it is required only for initial configuration of IP addresses.
- Cat 5 cables:
  These are standard cables used to connect LAN equipment and are terminated with RJ45 connectors. These are specified as either being standard or straight through or as cross over. Not recommended for speeds greater than 100 Mbps.
- Cat 5E (Cat 5 Enhanced) cables:
  The Cat 5E are the same as Cat 5 cables, but made to more stringent requirements. They are also designed for speeds up to 1 Gbps.
- Cat 6 cables:
  The same as Cat 5E, but made to more stringent standards. Designed for speeds up to 1 Gbps.
- Terminal server cables:
  Terminal server cables are a proprietary cable that can be used to interface between the MRV Terminal Server and various system components in order to allow terminal access.
- Twisted-pair telephone cables:
  These cables carry analog voice and digitized voice and data information between distribution frames and terminal devices throughout the building. They connect to 8-pin modular jacks located within 2.4 m (8 ft) of each device.

Consider cable length requirements and limitations for both initial installation and later growth when you plan a system.

Cable access

The customer is responsible for supplying all access for station, feeder, and riser cabling. This includes (where necessary):

- Conduit
- Floor boring
- Wall boring
- Access into hung ceilings

LAN design

Network requirements are critical to the CS 1000E quality of service. Ensure the network meets the following requirements:

- Provision 100BaseTx IP connectivity between the Call Server and the MG 1000E. The 100BaseTx IP connectivity can be either a point-to-point network or a distributed campus data network. IP daugherboards in the Call Server and the MG 1000E provide connectivity.
- Ensure that the 100BaseTx Layer 2 (or Layer 3) switch supports full-duplex connection. Routers are not supported in Call Server to MG 1000E connections. The ports on Layer 2 (or Layer3) switching equipment must be configured to autonegotiate ENABLED.
- Provision the ELAN subnet and the TLAN subnet on separate subnets.
- Provision all applications on the ELAN subnet on the same subnet. This includes Voice Gateway Media Cards that must be on the same ELAN subnet.
- Ensure that Voice Gateway Media Cards are in the same node on the same TLAN subnet.

For information on the requirements for creating a robust, redundant network, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Keep a record of the IP addresses assigned to system components. See Figure 49 "Sample IP address record sheet" (page 148) for a sample.

Figure 49
Sample IP address record sheet

xx.xx.xxx.128/25				
xx.xx.xxx.129	Gateway			
255.255.255.128	Subnet Mask	TLAN Subnet Mask	255.255.252.0	
xx.xx.xxx.128 – xx.xx.xxx.159	Logical blocks of 32 addresses (first four addresses in each block reserved for Layer 2 and Layer 3 equipment)	Node IP Address	xxx.xx.x.252	
xx.xx.xxx.160 – xx.xx.xxx.191		TLAN Gateway	xxx.xx.x.1	
xx.xx.xxx.192 – xx.xx.xxx.223				
xx.xx.xxx.224 – xx.xx.xxx.255				

ELAN Subnet Number	MAC address (by byte/octet)						Equipment Description	Serial Number	Comment	TLAN Subnet Address	Location	
	B1	B2	B3	B4	B5	B6					Rack	L/S/C
xx.xx.xxx.130	00	0C	F8	xx	xx	xx	Baystack 460	SDNIHR1xxx	ELAN_3		R2	
xx.xx.xxx.133	00	00	75	xx	xx	xx	SSC-0	NNTMG19XKVxx	MGT 01-0		R1	
xx.xx.xxx.134	00	20	D8	xx	xx	xx	Media Card-0	NNTMEJ02Bxxx	MGT 01-0	xxx.xx.x.30	R1	
xx.xx.xxx.135	00	00	75	xx	xx	xx	SSC-1	NNTMG19XKVxx	MGT 01-1		R1	
xx.xx.xxx.136	00	20	D8	xx	xx	xx	Media Card-1	NNTMEJ02Bxxx	MGT 01-1	xxx.xx.x.34	R1	
137-159												
xx.xx.xxx.165	00	00	75	xx	xx	xx	SSC-0	NNTMG19XKVxx	MGT 02-0		R4	
xx.xx.xxx.166	00	20	D8	xx	xx	xx	Media Card	NNTMEJ02Bxxx	MGT 02-0	xxx.xx.x.84	R4	
xx.xx.xxx.167	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	Branch Off	xxx.xx.x.75	R3	
xx.xx.xxx.168	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	H323_2	xxx.xx.x.70	R4	
xx.xx.xxx.169	00	00	75	xx	xx	xx	SSC	NNTMG19XKVxx	Branch Off		R3	
xx.xx.xxx.170	00	20	D8	xx	xx	xx	Media Card	NNTMEJ02Bxxx	Branch Off	xxx.xx.x	R3	
xx.xx.xxx.171	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	SIP_2	xxx.xx.x.37	R2	
xx.xx.xxx.172	00	00	75	xx	xx	xx	SSC-5	NNTMG19XKVxx	MGE 5		R2	8/0
xx.xx.xxx.173	00	00	75	xx	xx	xx	SSC-6	NNTMG19XKVxx	MGE 6		R2	8/1
174-191												
xx.xx.xxx.196	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	SIP/H323_1	xxx.xx.x	R1	
xx.xx.xxx.197	00	0C	F8	xx	xx	xx	Baystack 460	SDNIHR1xxx	ELAN_1		R2	
xx.xx.xxx.198	00	0C	F8	xx	xx	xx	Baystack 460	SDNIHR1xxx	ELAN_2		R2	
xx.xx.xxx.199	00	01	AF	xx	xx	xx	CPP-2 Core 0	NNTMxxxxxxxx	Core 0			
xx.xx.xxx.200	00	01	AF	xx	xx	xx	CPP-2 Core 1	NNTMxxxxxxxx	Core 1			
xx.xx.xxx.201	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	Sig Server 01	xxx.xx.x.18	R2	
xx.xx.xxx.202	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	NRS	xxx.xx.x.35	R2	
xx.xx.xxx.203	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	Gatekeeper	xxx.xx.x.119	R3	
xx.xx.xxx.204	00	20	D8	xx	xx	xx	Media Card	NNTMEJ02Bxxx	MGE 5	xxx.xx.x.136	R2	8/0/1
xx.xx.xxx.205	00	20	D8	xx	xx	xx	Media Card	NNTMEJ02Bxxx	MGE 6	xxx.xx.x.137	R2	8/1/1
206-223												
xx.xx.xxx.224	08	00	87	xx	xx	xx	Terminal Server	(same as MAC)	TS1	xx.xx.xxx.xx	R2	
xx.xx.xxx.225	08	00	87	xx	xx	xx	Terminal Server	(same as MAC)	TS2	none	R2	
xx.xx.xxx.226	00	0E	C0	xx	xx	xx	CallPilot	AC0xxxxx		xxx.xx.x.30		
xx.xx.xxx.227	00	0E	C0	xx	xx	xx	Symposium	ACxxxxxx	MGate: MGE x	xxx.xx.x.31		
xx.xx.xxx.228	00	0E	C0	xx	xx	xx	Baystack 470	SACC110xxx				
xx.xx.xxx.229	00	01	AF	xx	xx	xx	CPP-2 Core 0	NNTMxxxxxxxx	Geo-Red			
xx.xx.xxx.230	00	01	AF	xx	xx	xx	CPP-2 Core 1	NNTMxxxxxxxx	Geo-Red			
xx.xx.xxx.231	00	02	B3	xx	xx	xx	Sig Server	NNTM74XC0xxx	Geo-Red	xxx.xx.x.130		
							SS Node IP		Geo-Red	xxx.xx.x.129		
							IP Phone		Geo-Red	xxx.xx.x.131		
							IP Phone		Geo-Red	xxx.xx.x.132		
232-255												
Legend: L/S/C = Loop, Shelf, Card; SSC = Small System Controller, Sig Server = Signaling Server												

Preparing a floor plan

Prepare a detailed floor plan for each site. The floor plan must indicate the size and location of:

- the racks, including planned expansion areas
- the service panel
- system terminal, printer, or other terminal devices (such as modems)
- PTFUs (if equipped)
- space for additional equipment, such as reserve power equipment or auxiliary processors

> **ATTENTION**
>
> **IMPORTANT!**
>
> According to the National Fire Code, equipment must be at least 30.5 cm (12 in.) from a sprinkler head.

Ensure that the site configuration meets all requirements of the third-party suppliers of the 19-inch racks.

Creating a building cable plan

To create a building cable plan, complete the following tasks.

1. Show the routing of all wiring, the location and wiring requirements of each terminal device connected to the system, and any other relevant information about the device.
2. Show the location of distribution frames, conduits and access points, and power outlets.
3. Identify the ownership of existing building wire if it is to be used.
4. Perform a random sampling of in-place wiring to ensure that it meets specifications for high-speed lines. All wiring carrying high-speed data must pass a verification test as part of the installation procedures.
5. Identify the location of conduits and floor ducts. If telephone cable is run in conduit then that conduit cannot be used for any other wiring.
6. Provide three pairs of telephone wire from a distribution frame to a nearby telephone jack for each terminal device. Modular jacks must be within 2.0 m (8 ft) of the device.
7. Provide Power over LAN cables to all desktops.
8. Divide the building cable plan into zones. Zones are typically the termination point of conduits throughout the office. Identify each zone on the building cable plan with a letter or number, and assign a block of numbers to each zone. Figure 50 "Building cable zones" (page 152) illustrates zoning.

Be sure to leave room for expansion.

Wire routing

Refer to the appropriate electrical code for your region for standards you are required to meet. For the US, refer to the National Electrical Code (NEC).

To plan wire routing, establish the start and end point of each cable relative to the location of the terminal devices in the building, and then examine the construction of the office to determine the best wiring routes. Consider the following guidelines when performing this task.

- Floors:
  - In the open, wires can run along baseboard, ceiling moldings, or door and window casings. For the safety of employees, never run wire across the top of the floor.
  - When concealed, wires can run inside floor conduits that travel between distribution frames and jacks. (Under-carpet cable is not recommended.)
- Ceilings:

  National and local building codes specify the types of telephone wire that you can run in each type of ceiling. Local building codes take precedence.
- Walls:

  Cables that run vertically should, when possible, run inside a wall, pole, or similar facility for vertical wire drops. Cables that run horizontally cannot be blind-fed through walls.
- Between floors:

  Locate distribution frames as closely to one another as possible. Local coding laws specify whether or not a licensed contractor is required if conduit is installed.
- EMI:

  Data degradation can occur if wires travel near strong EMI sources. See "Electromagnetic and radio frequency interference" (page 144) for a description of common interference sources.

Termination points

After you determine the wire routing, establish termination points. Cables can terminate at:

- the MDF (typically in the equipment room)
- intermediate distribution frames, typically on each floor in telephone utility closets

- wall jacks to terminal boxes, typically located near the terminal device

At the distribution frame (also called the cross-connect terminal), house cables terminate on the vertical side of the two-sided frame and cross connect to equipment that is typically located on the horizontal. If you use a color field scheme, house cables typically terminate in the blue field and the equipment terminates on the purple (US) or white (Canada) field.

In all cases, clearly designate the block where the cables terminate with the cable location information and the cable pair assignments. Keep a log book (cable record) of termination information. See Figure 51 "Sample cable record" (page 153) for an example.

Figure 50
Building cable zones

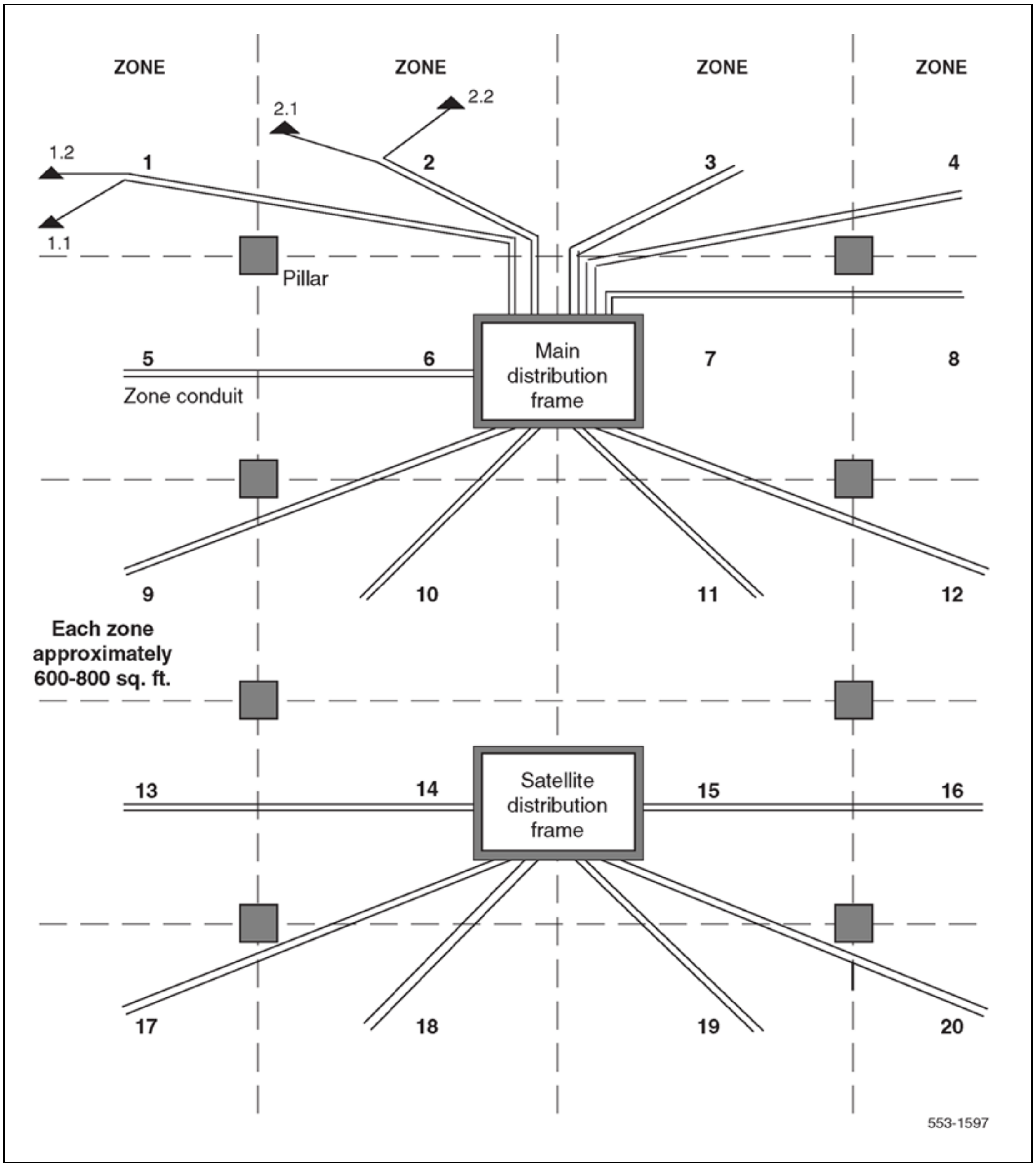


Figure 51
Sample cable record

CABLE RECORD

Customer ______________________________

Location ______________________________

Cable ____________________ Binder __________ Page ____ of ____

DN	TN M	TN S	TN C	TN U	NAME	FEATURES / REMARKS	TERMINAL DEVICE	BLOCKS DF / HOUSE	COLOR
									W BL
									W OR
									W GR
									W BR
									W SL
									R BL
									R OR
									R GR
									R BR
									R SL
									BK BL
									BK OR
									BK GR
									BK BR
									BK SL
									Y BL
									Y OR
									Y GR
									Y BR
									Y SL
									V BL
									V OR
									V GR
									V BR
									V SL

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Preparing for delivery

When preparing for equipment delivery, answer the following questions:

- Has a request been made for equipment delivery?
- Are transportation arrangements to the premises completed?
- Is a list of all ordered equipment available on site?
- Is help needed and available for preparing the equipment room?
- Are unloading and unpacking facilities and tools available?
- Is help needed and available for delivery?

Plan to unload equipment as close to the final installation area as possible for an easier, and perhaps safer, installation.

Conducting pre-installation inspections

Obtain any appropriate sign-off before the site is ready for equipment delivery and installation. Sign-off can include regulatory items such as electrical inspections, air conditioning inspections, and cable plan approval. In addition, an overall equipment room inspection and a building cable inspection should be performed before installation.

- Inspect the room to verify that:
  - All physical and environmental requirements are met.
  - System grounding and power equipment is installed and tested.
  - The equipment layout is marked on the floor.
- Inspect building cable to verify that:
  - Sufficient distribution frames are provided.
  - Conduits or floor ducts to terminal locations are installed.
  - Terminal jacks are installed.
  - Sufficient wiring is available.

Inspect the equipment room to verify that all physical and environmental requirements are met, system grounding and power equipment is installed and tested, and the equipment layout is marked on the floor.

Inspect the building cable to verify that sufficient distribution frames are provided, conduits or floor ducts to terminal locations are installed, terminal jacks are installed, and sufficient wiring is on hand.

Preparing for installation

The installation plan, work orders, and appropriate documentation should be on hand at the time of installation.

Reviewing the installation plan

The installation plan can consist of the equipment room floor plan, the building cable plan, and an installation and test sequence chart.

The equipment room floor plan should show:

- Racks, including planned expansion areas
- Main distribution frame
- Service panel
- System terminal, printer, or other terminal devices
- External power equipment (such as UPS)
- Cable racks
- PFTUs (if equipped)

The building cable plan should show:

- Cable routing and designation information
- Location of each terminal device
- Type of cable or wiring required for each terminal device
- Location of all distribution frames and system and terminal cross-connect assignments
- Location of conduits and floor ducts, including access points
- Location of power outlets for terminal devices

An installation and test sequence (ITS) chart shows typical installation tasks, the sequence of the tasks, and task start and duration information.

Reviewing the work orders

The work order can include:

- Detailed listing of the equipment ordered
- Terminal Number (TN) assignments
- Directory Number (DN) assignments for each terminal device
- IP assignments for all equipment
- Office Data Administration System (ODAS) designators for each terminal device (if the software package is equipped)
- Features available to each telephone and data set
- Administration database entries for telephone and data set features

Reviewing the documentation

Instructions for unloading and unpacking system equipment, as well as a full set of standard Nortel technical publications (NTPs), are delivered with each system.

Power and grounding

Contents

This section contains information on the following topics:

"Introduction" (page 157)

"Grounding requirements" (page 157)

"Grounding methods" (page 162)

"Commercial power requirements" (page 165)

"Alternative AC-powered installation" (page 166)

"AC input requirements" (page 168)

"Power consumption" (page 168)

"Heat dissipation" (page 173)

"Uninterruptible Power Supply" (page 174)

"Power requirements for IP Phones" (page 175)

Introduction

CS 1000E system components are AC-powered. This section outlines the system's grounding and electrical requirements.

Grounding requirements

For system grounding in new installations, Nortel recommends following ANSI/TIA/EIA-607 (Commercial Building and Bonding Requirements for Telecommunications Equipment).

In building installations where the ANSI/TIA/EIA-607 method is not used, connect the equipment ground to the AC ground at the respective service panel.

If you are having difficulty interpreting the grounding methods in this document, Nortel recommends obtaining the services of a certified power contractor or auditor prior to system installation or cutover

WARNING
Failure to follow grounding recommendations can result in a system installation that is:

- unsafe for personnel handling or using the equipment
- not properly protected from lightning or power transients
- subject to service interruptions

Before installing the equipment and applying AC power, measure the impedance of the building ground reference. An ECOS 1023 POW-R-MATE or similar meter is acceptable for this purpose. Ensure that the ground path connected to the system has an impedance of 4 ohms or less. Make any improvements to the grounding system before attempting installation.

DANGER
DANGER OF ELECTRIC SHOCK
Never connect the single point ground conductor from the system to structural steel members or electrical conduit. Never tie this conductor to a ground source or grounded electrode that is not hard-wired to the building reference conductor.

System grounding must adhere to the following requirements:

- The ground path must have an impedance of 4 ohms or less.
- Ground conductors must be at least #6 AWG (16 mm^2) at any point (see Table 13 "Area-specific ground wire requirements" (page 159) for a list of grounding wire requirements specific to some areas).
- Ground conductors must not carry current under normal operating conditions.
- Spliced conductors must not be used. Continuous conductors have lower impedance and are more reliable.
- All conductors must terminate in a permanent way. Make sure all terminations are easily visible and available for maintenance purposes.

- Tag ground connections with a clear message such as "CRITICAL CONNECTION: DO NOT REMOVE OR DISCONNECT."

Table 13
Area-specific ground wire requirements

Area	Ground wire requirements
Germany	#8 AWG (10 mm^2) green/yellow wire
Other areas in Europe	Not smaller than #6 AWG (16 mm^2) at any point
UK	Two green/yellow wires no thinner than 10 mm^2

For more information about standards and guidelines for grounding telecommunications equipment, refer to ANSI/TIA/EIA-607 (Commercial Building and Bonding Requirements for Telecommunications Equipment).

DANGER
DANGER OF ELECTRIC SHOCK
For an installed Call Server, Media Gateway, Media Gateway Expander, or Signaling Server, link impedance between the ground post of any equipment and the single point ground that it connects to must be less than 0.25 ohms.

CAUTION
Damage to Equipment
Transients in supply conductors and ground systems can damage integrated circuits. This damage can result in unreliable system operation. Damage caused by transients is not always immediately apparent. Degradation can occur over a period of time.

Single Point Ground

Correct grounding of communications systems is necessary to protect equipment from the hazards of surge and noise interference. The Single Point Ground (SPG) method of protecting communications equipment is the Nortel standard. Table 14 "Grounding design considerations" (page 159) describes grounding design considerations.

Table 14
Grounding design considerations

Safety	• Dissipate unwanted surge energies such as lightning striking on the outside plant

	• Fuses or breakers open to disrupt the excessive current flow caused by a power fault
Equipment protection	• Grounding for outside plant cable shields and protectors • Grounds for framework and logic references
Electromagnetic compatibility (EMC)	• Conform with electromagnetic compatibility (EMC) grounding requirements
Installation and maintenance	• Cost-effective to install and maintain when part of the initial electrical installation • Correcting violations of national codes after the initial installation is difficult and costly
Powering	• If the equipment is backed up with an Uninterruptable Power Supply (UPS), consider the grounding of all equipment that is part of the telecommunications system as a single system
Advances in technology	• Provides important protection to ensure the effective operation of circuit cards and avoid costly downtime and replacement

DANGER
DANGER OF ELECTRIC SHOCK
Do not perform work inside electrical panels unless you are a qualified electrician. Do not try to remove bonding conductors without approval from qualified personnel.

In an ANSI/TIA/EIA-607 installation, the Telecommunications Main Grounding Busbar (TMGB)/Telecommunications Grounding Busbar (TGB) links the telecommunications equipment to the ground. Other grounding terminology is:

- building principal ground, normally in a building with one floor
- floor ground bar, normally in buildings with more than one floor

Configure telecommunications subsystems, such as groups of frames or equipment, as separate single-point ground entities connected to the equipment's dedicated service panel via a single-point ground bar. The service panel ground connects to the building principal ground via the main service panel or, in an ANSI/TIA/EIA-607 installation, via the TGB. Refer to Figure 53 "Typical wiring plan" (page 162).

Figure 52
ANSI/TIA/EIA-607 grounding schematic

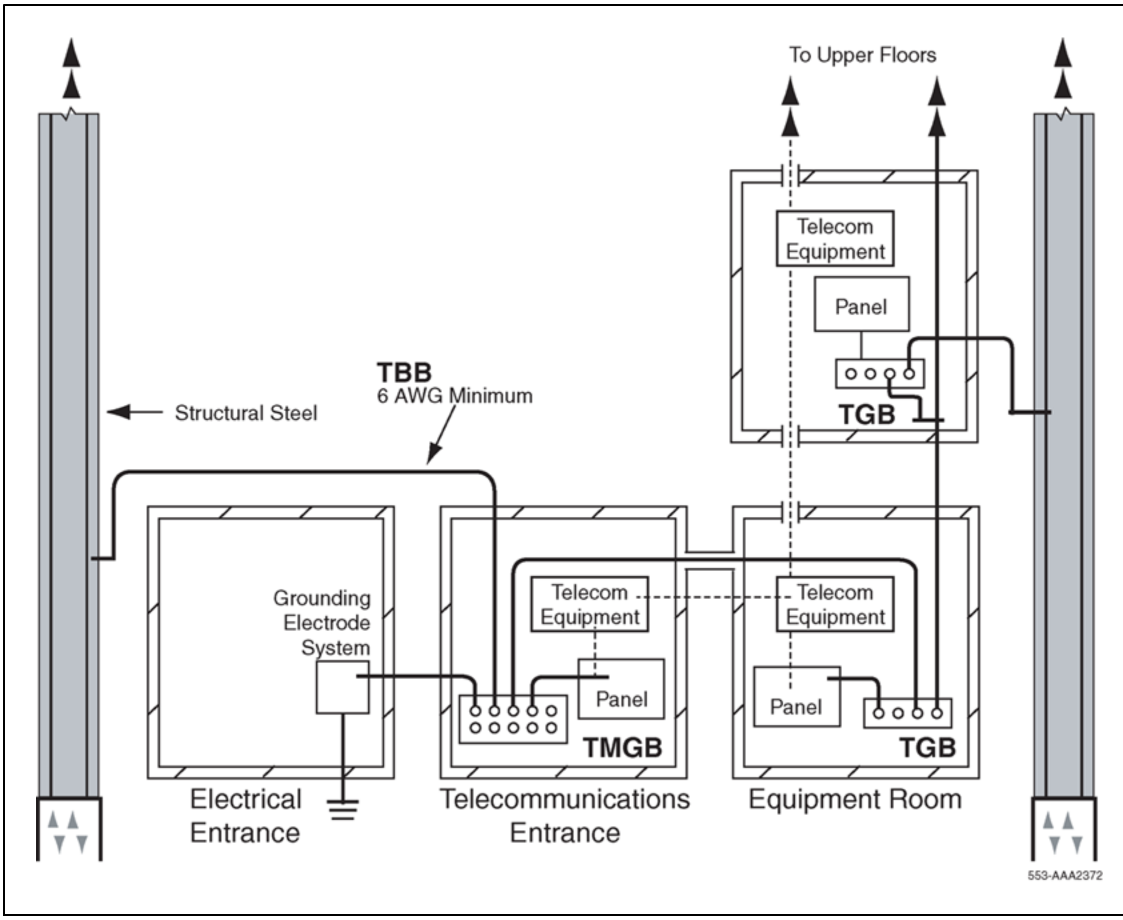
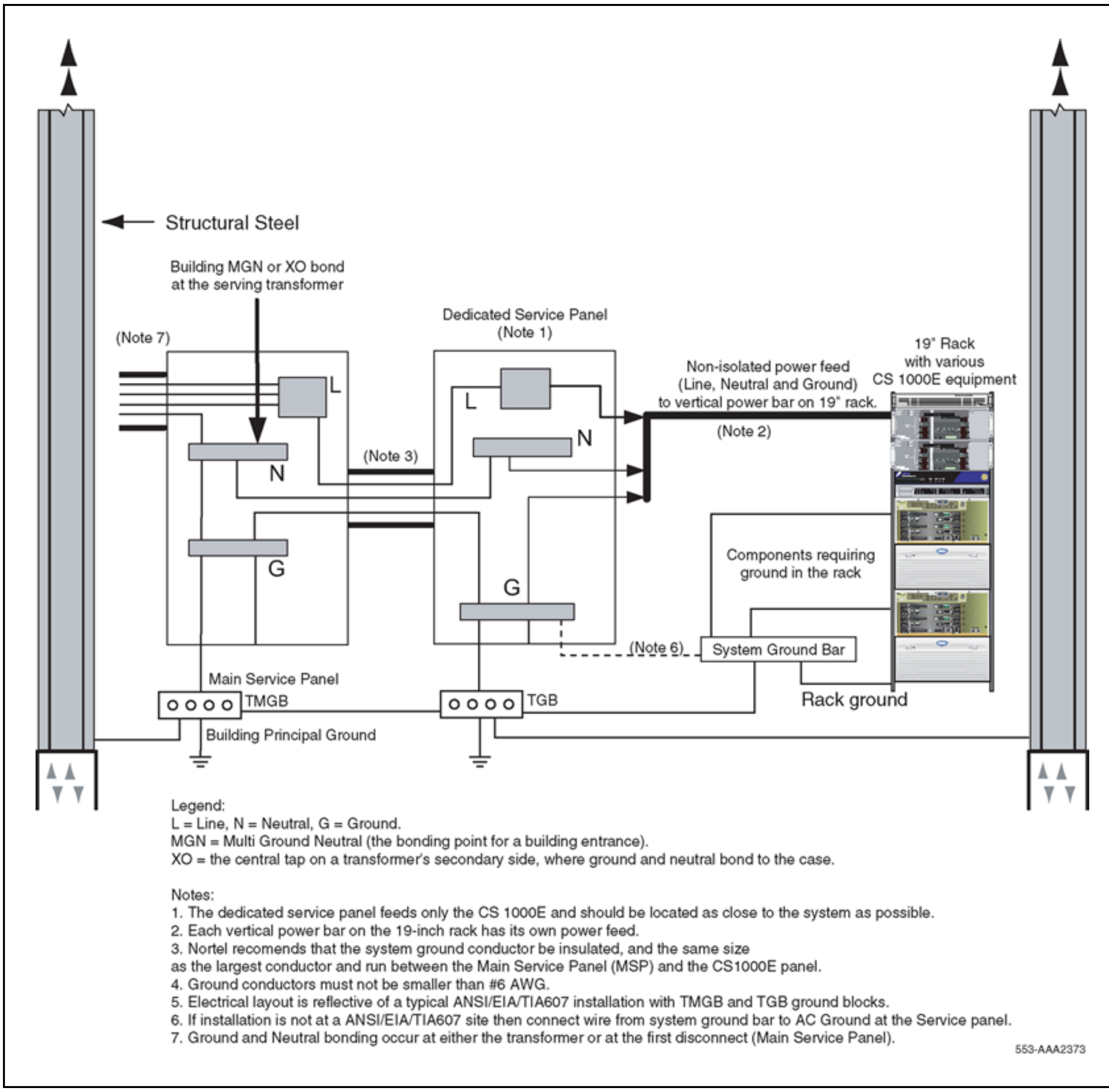


Figure 53
Typical wiring plan



Grounding methods

This section describes the grounding methods for:

- "Ground bar (NTBK80)" (page 163)
- "Ground bar (NTDU6201)" (page 163)
- "CS 1000E" (page 163)
- "Signaling Server" (page 163)
- "Media Gateway" (page 164)
- "Media Gateway Expander" (page 164)
- "Multiple components in a rack" (page 164)

DANGER
DANGER OF ELECTRIC SHOCK
To prevent ground loops, power all CS 1000E system equipment from the same dedicated power panel.

Ground bar (NTBK80)

The NTBK80 ground bar is capable of grounding up to six Media Gateways (either with or without companion Media Gateway Expanders) back to the SPG. See Table 13 "Area-specific ground wire requirements" (page 159) for area-specific ground wire requirements.

Ground bar (NTDU6201)

If there are more than six Media Gateways (either with or without companion Media Gateway Expanders), use the NTDU6201 ground bar. The NTDU6201 can be adjusted for various mounting configurations. It accepts 35 #6 AWG (16 mm^2) wire connections. The ground bar must terminate at the service panel ground. See Figure 53 "Typical wiring plan" (page 162).

CS 1000E

The CS 1000E does not connect to a ground bar. It is properly grounded when:

- The CS 1000E power cord is plugged into the rack's AC outlet. The rack's AC outlet must be grounded to its dedicated electrical panel.
- The CS 1000E power cord is plugged into a wall AC outlet. The CS 1000E is grounded outside of the rack via the safety grounding conductor in the power cord. This method only ensures proper grounding of the CS 1000E itself. It does not provide grounding protection for other rack-mounted pieces of equipment. Therefore, ensure that other devices in the rack are properly grounded as required.

Signaling Server

The Signaling Server does not connect to a ground bar. It is properly grounded when:

- The Signaling Server power cord is plugged into the rack's AC outlet. The rack's AC outlet must be grounded to its dedicated electrical panel.
- The Signaling Server power cord is plugged into a wall AC outlet. The Signaling Server is grounded outside of the rack via the safety grounding conductor in the power cord. This method only ensures proper grounding of the Signaling Server itself. It does not provide grounding protection for other rack-mounted pieces of equipment. Therefore, ensure that other devices in the rack are properly grounded as required.

Media Gateway

The grounding method used for the Media Gateway depends on the number of units used and whether the units are powered by the same service panel.

All equipment located in a series of equipment racks that are physically bonded together must be grounded to and powered by the same service panel. If additional service panels are required, collocate them beside the original service panel.

If racks are not bonded together, then the equipment located in the racks can be grounded and powered by separate service panels.

Connect a #6 AWG (16 mm^2) ground wire from the rear panel grounding lug of each Media Gateway to the ground bar. See Table 13 "Area-specific ground wire requirements" (page 159) for area-specific ground wire requirements. Connect the ground bar to a ground source in the dedicated service panel.

In the UK, connect the ground wire from the equipment to a ground bar or through a Krone Test Jack Frame.

Media Gateway Expander

The Media Gateway and Media Gateway Expander are considered as the same ground. To ground the Media Gateway Expander, jumper the ground wire from it to the grounded Media Gateway.

ATTENTION

IMPORTANT!
Power each Media Gateway and Media Gateway Expander pair from the same service panel.

Multiple components in a rack

To ground multiple pieces of equipment installed in a rack, run a separate connection from the grounding lug on each piece of equipment to the ground bar.

If a piece of equipment in a rack does not have a grounding lug, ground the rack to the ground bar. Grounding the rack in this manner grounds the equipment by the SPG method.

Conduit requirements

Conductive conduit linking panels and equipment is legal for use as a grounding network in most countries. For all CS 1000E system ground paths, route the correct size of insulated copper conductors inside conduit. A ground link that depends on a conduit can defeat the improvements achieved with the installation of dedicated electrical panels and transformers. A grounding failure can result from the following:

- Personnel who service different equipment can separate conduit links. If such a separation occurs between the system and the building ground reference, the conduit cannot provide a ground path. This situation is hazardous.
- Corrosion of metal conduits increases resistance. Threaded connections are prone to corrosion. This problem becomes worse when there are multiple links. Applying paint over the conduit increases the corrosion process.
- Conduit cannot be fastened to secure surfaces. Often, the conduit bolts on to structural steel members, which can function as ground conductors to noisy equipment (for example, compressors and motors). Adding noisy equipment into the grounding system can damage the system’s performance. The resulting intermittent malfunctions can be difficult to trace.

Commercial power requirements

The CS 1000E system is AC-powered. Optimally, a dedicated electrical panel that is connected to the facility’s electrical system powers the system. The dedicated electrical panel provides power only to the CS 1000E system components and its related telecommunications hardware such as TTYs and printers. There is no expectation that system components that are located off-site will be powered by this dedicated electrical panel.

Media Gateway and Expander

Each Media Gateway-Expander pair must share the same electrical breaker and outlet. Refer to "AC input requirements" (page 168) for more information.

If the system is equipped with CallPilot, the CallPilot server must connect to the same dedicated service panel that feeds the MG 1000E that the MGate card resides.

Rack power bars

Power each power bar or rack-mounted power rail on a separate circuit fed from the service panel.

The rating for power bars must be 120 or 240 V, 15 or 20 A, 50-60 Hz, grounded. Power bars are non-isolated ground type.

Powering redundant equipment

Provide power to redundant equipment from dedicated power bars fed from their own dedicated circuits.

Powering auxiliary equipment

Terminals, printers, modems, and other data units used with the CS 1000E or MG 1000T have special wiring requirements. All equipment must be:

- powered from the same electrical panel or transformer as the system
- grounded to the same electrical panel or transformer as the system
- labeled at the electrical panel to prevent a non-authorized power interruption

Alternative AC-powered installation

Use an approved isolation transformer when the power to all system components at a location cannot be supplied by a dedicated electrical panel or the dedicated electrical panel cannot provide optimal conditions. Refer to Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) and Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) to determine the system power consumption.

The isolation transformer must have the following characteristics:

- 120/240 V AC input, over-current protected at primary.
- 120/240 V AC available at secondary outputs, each protected by circuit breaker.
- Primary and secondary windings completely isolated from one another.
- Approved for use locally as a stand-alone user product (CSA, UL, or other locally recognized clear markings).
- Capable of providing power to all components operating at the same time at full load.
- Equipment unrelated to the system must not be powered from a transformer that provides service to the system.

Installing an isolation transformer ground

The transformer ground must have separate grounds for primary and secondary windings rather than a common ground. Ground conductors inside the transformer must be correctly sized.

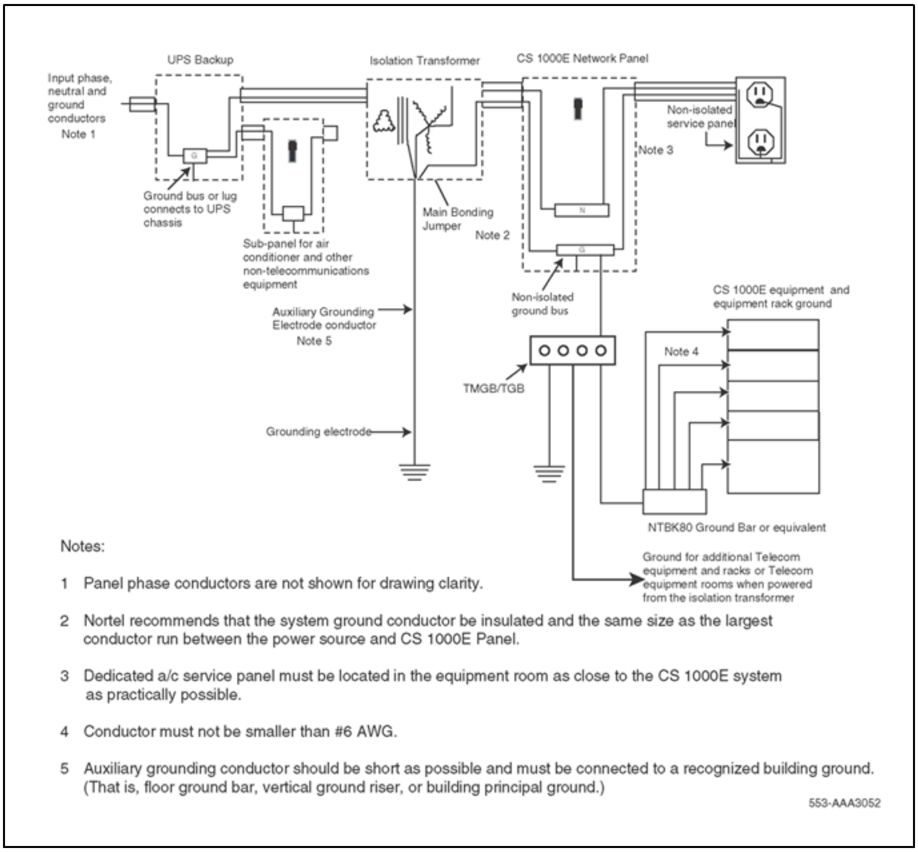
DANGER
DANGER OF ELECTRIC SHOCK
Nortel does not recommend connecting any CS 1000E system telecommunications ground bus to untested horizontal structural steel or water pipes, or other unreliable ground paths. Use a ground point known to be "clean" and permanent. Place a "DO NOT DISCONNECT" tag on it.

Procedure 1 "Installing an isolation transformer without pluggable power cords" (page 167) describes the method to install an isolation transformer without pluggable power cords.

Procedure 1
Installing an isolation transformer without pluggable power cords

Step	Action
1	If the transformer does not have a pluggable cord, hardwire the transformer to an electrical panel. Route all wires (including grounds) through a single conduit. Some electrical codes permit the use of conduit as the only ground conductor between pieces of equipment.
2	Run a separate insulated ground conductor through the conduit to hold unit grounds together. Such a conductor maintains the safety ground connection in the event that the conduit becomes corroded or disconnected.
3	Run all ground lines through the same conduit as the phase conductors that serve the equipment. Figure 54 "Typical hardwired isolation transformer wiring plan" (page 168) shows the isolation transformer connections.

Figure 54
Typical hardwired isolation transformer wiring plan



—End—

AC input requirements

Refer to Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) for the AC input requirements of CS 1000E components.

If other data communications equipment is in the same rack as the CS 1000E system, power each piece of equipment from the same electrical panel. Install additional outlets, if necessary.

Because local power specifications vary, consult a qualified local electrician when planning power requirements.

Power consumption

System power consumption depends on the number of components installed.

Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) summarizes the current, power, and cooling requirements for CS 1000E components. Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) shows absolute maximum ratings as well as typical ratings for configured systems. The typical values are provided as a guide to avoid over-engineering, particularly for Uninterruptable Power Supply (UPS) requirements.

Table 15
Current, power, and cooling requirements for CS 1000E components

	Current @ 120/240 V AC (A)		Required UPS power (W)		Thermal dissipation (Btu)	
Component	**Maximum**	**Typical**	**Maximum**	**Typical**	**Maximum**	**Typical**
NTDU62 Call Server	2.50/1.25	1.00/0 .50	300.00	120.00	1023.90	409.56
NTDU27 (Intel ISP1100)	2.00/0.90	0.50/0 .25	200.00	60.00	682.60	204.78
NTDU97 (HP DL320 G4)	6.00/3.0	4.0/2.0	580.00	400.00	1990.00	1370.00
NTDU99 (IBM x306m)	5.50/2.8	3.00/1.5	550.00	350.00	1024.00	682.00
NTDU14 Media Gateway	1.40/0.70	1.17/0 .58	300.00	190.00	1023.60	648.30
NTDU15 Media Gateway Expander	1.15/0.58	1.17/0 .58	300.00	145.00	1023.60	494.70
MRV Terminal Server	1.60/0.80	0.40/0 .20	192.00	48.00	655.30	163.83
BayStack 470	1.50/0.75	0.60/0 .30	90.00	72.00	324.00	245.74

Maximum values for the Media Gateway and Expander assume worst case conditions. It is difficult to specify a typical configuration. The typical values in the table are intended as a rough guide for quick estimations. Nortel recommends that qualified personnel take current measurements for a more accurate assessment of UPS and thermal requirements.

Maximum AC input for the BayStack 460 includes maximum power of the Power over LAN. The typical rating has been adjusted to reflect configuring for IP Phones (60 mA at 48 V DC).

Maximum voltage limits: North America – 90 and 132 V, single phase. Europe and UK – 180 and 250 V, single phase. Frequency: North America – 60 Hz. Europe and UK – 50 Hz. Fuse: Germany – 16 A.

Component	Current @ 120/240 V AC (A)		Required UPS power (W)		Thermal dissipation (Btu)	
	Maximum	Typical	Maximum	Typical	Maximum	Typical
BayStack 460 (Power over LAN not used)	4.70/2.40	0.60/0.30	295.00	72.00	335.00	245.74
BayStack 460 (Power over LAN for 24 IP Phones)	4.70/2.40	1.20/0.60	364.12	141.12	335.00	245.74

Maximum values for the Media Gateway and Expander assume worst case conditions. It is difficult to specify a typical configuration. The typical values in the table are intended as a rough guide for quick estimations. Nortel recommends that qualified personnel take current measurements for a more accurate assessment of UPS and thermal requirements.

Maximum AC input for the BayStack 460 includes maximum power of the Power over LAN. The typical rating has been adjusted to reflect configuring for IP Phones (60 mA at 48 V DC).

Maximum voltage limits: North America – 90 and 132 V, single phase. Europe and UK – 180 and 250 V, single phase. Frequency: North America – 60 Hz. Europe and UK – 50 Hz. Fuse: Germany – 16 A.

Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) provides the power consumption, UPS power requirements, and thermal dissipation of Media Gateway packs (circuit cards and daughterboards) commonly installed in CS 1000E and MG 1000E Media Gateways and Expanders.

Electrical load for analog line cards varies with traffic load. The figures in Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) assume that 50% of analog lines are active. The UPS power wattage figures also take into account the average efficiency of the Media Gateway power supplies.

Table 16
Power and cooling requirements for Media Gateway packs

Media Gateway pack	Active off-hook (%)	Power consumption (W)	UPS power (W)	Thermal dissipation	
				W	Btu
NTDW60 Media Gateway Controller card	N/A	7	7	7	24
NTDW61/66 CP PM	N/A	30	30	30	102
NTDW62 MGC DSP daughterboard (32 port)	N/A	4	4	4	14

Media Gateway pack	Active off-hook (%)	Power consumption (W)	UPS power (W)	Thermal dissip ation	
				W	Btu
NTDW64 MGC DSP daughterboard (64 port)	N/A	4	4	4	14
NTDK20 Small System Controller card	N/A	16	24.0	24.0	81.9
NTDK83 100BaseT daughterboard (dual-port)	N/A	6	9.0	9.0	30.7
NTDK99 100BaseT daughterboard (single-port)	N/A	4	6.0	6.0	20.5
NT5K02 Flexible Analog Line card	50	26	39.0	6.6	22.5
NT8D02 Digital Line card	100	26	39.0	13.0	44.4
NT8D03 Analog Line card	50	26	39.0	6.6	22.5
NT8D09 Analog Message Waiting Line card	50	26	39.0	6.6	22.5
NT8D14 Universal Trunk card	DID-enab led	28	42.0	42.0	143.3
NT8D15 E&M Trunk card	N/A	29	43.5	43.5	148.4
NTAK09 1.5MByte DTI/PRI card	N/A	10	15.0	15.0	51.2
NTAK10 2.0 MByte DTI card	N/A	12	18.0	18.0	61.4
NTAK79 2.0 MByte PRI card	N/A	12	18.0	18.0	61.4
NTBK50 2.0 MByte PRI card	N/A	12	18.0	18.0	61.4
NTRB21 TMDI (1.5 Mbyte DTI/PRI) card	N/A	12	18.0	18.0	61.4
NTVQ01 Media Card (32 port)	N/A	18	27.0	27.0	92.1
NTDW65 MC32S Media Card (32 port)	N/A	9	9	9	31

ATTENTION

The UPS power requirement is the card’s power consumption divided by the efficiency factor for the Media Gateway power supply plus peak inrush. For Media Gateways use 1.5 times the wattage to give the UPS wattage, or volt-amps (VA).

For digital and analog (500/2500-type) telephones, most thermal dissipation will be external to the switch room. In Table 16 "Power and cooling requirements for Media Gateway packs" (page 170), thermal dissipation values for these cards have been adjusted accordingly.

Power consumption worksheets

Table 17 "CS 1000E system power consumption worksheet" (page 172) and Table 18 "Media Gateway power consumption worksheet" (page 173) provide worksheets to determine power consumption for the CS 1000E system.

Prepare one worksheet for the system as a whole (Table 17 "CS 1000E system power consumption worksheet" (page 172)).

Table 17
CS 1000E system power consumption worksheet

Component		Required UPS power		Thermal dissipation	
	Number of comp. (1)	Per comp. (W) (2)	Total (W) (1) x (2)	Per comp. (Btu) (3)	Total (Btu) (1) x (3)
Call Server					
Signaling Server					
Terminal Server					
BayStack 470					
BayStack 460 (Power over LAN not used)					
BayStack 460 (Power over LAN for 24 IP Phones)					
NTDU14 Media Gateways*					
NTDU15 Media Gateway Expanders*					
TOTAL					
*Enter the sum of the totals for individual Media Gateways and Expanders (Table 18 "Media Gateway power consumption worksheet" (page 173)).					

Prepare one worksheet for each Media Gateway and Media Gateway Expander, if equipped (Table 18 "Media Gateway power consumption worksheet" (page 173)). Refer to Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) for the power and thermal dissipation requirements for the individual components.

Table 18
Media Gateway power consumption worksheet

Media Gateway number ________________				
Slot	**Media Gateway Pack**	**Required UPS power (W)**	**Thermal dissipation**	
			(W)	**(Btu)**
0	Media Gateway Controller or Small System Controller			
0	DSP daughterboard (32 port)			
0	DSP daughterboard (64 port)			
1				
2				
3				
4				
Media Gateway Expander				
7				
8				
9				
10				
TOTAL				

Heat dissipation

The CS 1000E is equipped with a cooling system and does not have heat dissipation problems under normal applications. Mounting in the rack is not restricted.

See Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) and Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) for the thermal load generated by system components and Media Gateway packs.

For air conditioning purposes, 1 ton = 12 000 Btu.

Uninterruptible Power Supply

An Uninterruptible Power Supply (UPS) generally consists of a combination battery charger (AC/DC converter) and inverter (DC/AC converter), along with associated batteries. The batteries can be internal or external to the UPS. A UPS is not a standby power source, but an online unit with no output interruption when the AC power is interrupted.

There are a number of UPS vendors and systems available. Factors to consider when choosing a UPS include:

- input voltage and frequency range
- output voltage and current capacity
- number and type of output receptacles
- regulatory and safety agency approvals
- efficiency and performance considerations
- alarm and status indications
- battery recharge time
- the maximum time backup power is required
- existing batteries or other power equipment available at the site
- future system growth

UPS sizing

To determine UPS sizing, sum the values given in Table 15 "Current, power, and cooling requirements for CS 1000E components" (page 169) and Table 16 "Power and cooling requirements for Media Gateway packs" (page 170) for UPS requirements for the applicable components and Media Gateway packs. The value in watts (W) is equivalent to a volt-ampere (VA) rating. Size the UPS in terms of its rating in VA (or kVA). For AC-powered systems, EC calculates the system power consumption in both watts and volt-amperes.

To determine the sizing and provisioning of UPS batteries, follow the instructions provided by the UPS manufacturer. A general approach is to take the total system power in watts, divide by the UPS inverter efficiency, and convert to battery current drain by dividing by the nominal discharge voltage of the battery string. Then determine the battery requirements in ampere-hours (A-hrs) by multiplying the battery current drain by the required reserve power operating time.

$$Ahr = \left(\frac{W_{total}}{V_{dischg}}\right)T_{reserve}$$

UPS installation

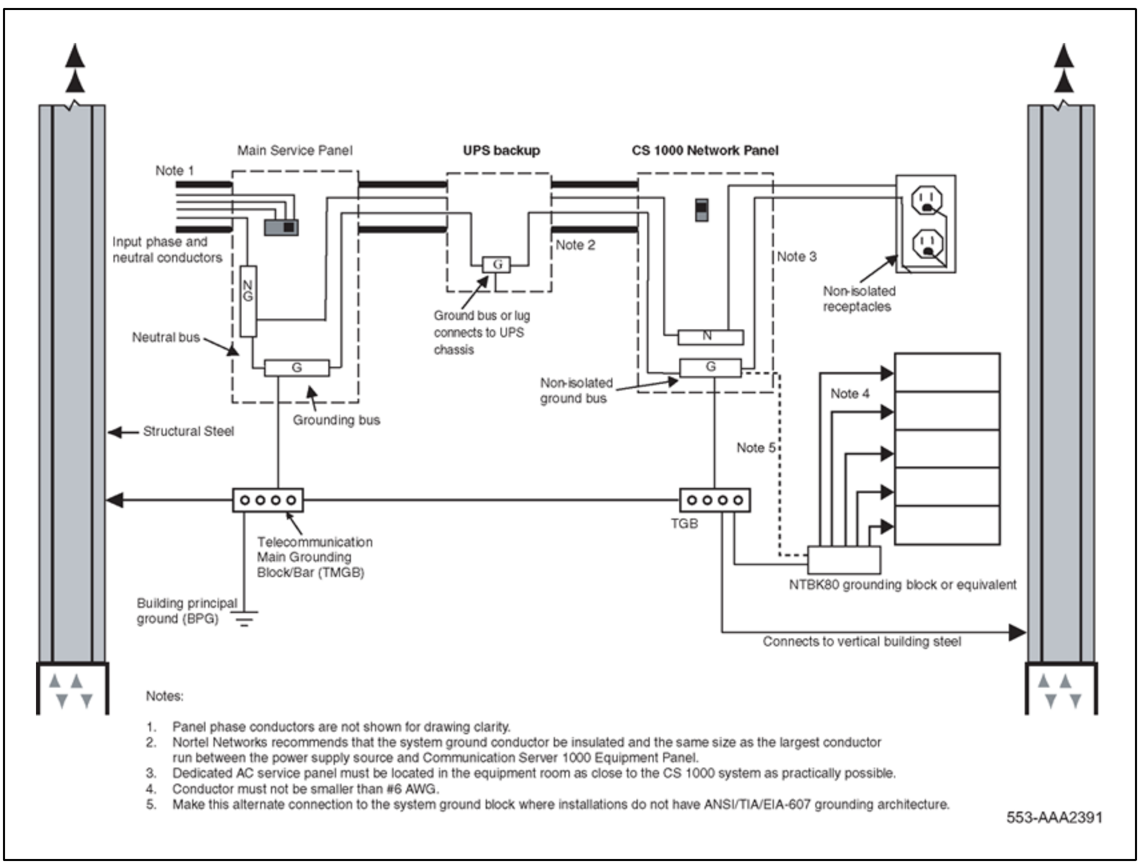
When installing a UPS, follow the vendor’s instructions carefully.

Nortel recommends installing a bypass switch during the initial UPS wiring (if the switch function is not inherently a part of the UPS itself). The UPS bypass switch lets the system run directly from the commercial power source while the UPS is taken off-line during installation, service, or battery maintenance.

> **CAUTION**
> **Damage to Equipment**
> Take care when connecting battery cables to the UPS. Connecting battery cables backward can result in severe damage to the UPS.

Figure 55 "Typical UPS wiring plan" (page 175) shows a typical UPS wiring plan.

Figure 55
Typical UPS wiring plan



Power requirements for IP Phones

The IP Phones and IP Softphone require 16 V AC, 500 mA that is supplied by a local transformer. The appropriate transformer depends on the line voltage, which is different for each country.

IP Phones also accommodate 48 V DC power. IP Phones can be powered over the LAN by a Layer 2 switch such as a BayStack 460 (see "Layer 2 switch" (page 78)). For more information about Power over LAN, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Design parameters

Contents

This section contains information on the following topics:

"Introduction" (page 177)

"System parameters" (page 178)

"Customer parameters" (page 178)

"Console and telephone parameters" (page 179)

"Trunk and route parameters" (page 180)

"ACD feature parameters" (page 181)

"Special feature parameters" (page 181)

"Hardware and capacity parameters" (page 182)

"Memory-related parameters" (page 184)

Introduction

This section describes sets of design parameters that set an upper boundary on certain system capacities. Changes to these parameters generally require a revision to the software and are constrained by other basic capacities such as memory and traffic or system load. The design parameters are set to provide the best possible balance between limits.

Note on terminology

Each Media Gateway in the CS 1000E can be connected to an optional Media Gateway Expander in order to increase the number of card slots available. In this chapter, all references to Media Gateways include the optional Media Gateway Expander, if equipped.

System parameters

Table 19 "System parameters" (page 178) lists system parameters and provides their maximum values.

Table 19
System parameters

System parameters	Maximum value	Comments
Customers	100	
Display messages for background terminal	255	
Input/output ports (for example, TTYs and printers)	16	Two physical (Com 1 and Com 2) and fourteen PTYs to Terminal Server (history file counts as one device)
AML/CSL links	16	IP links
TNs – CS 1000E	32 676	Software design limit. Actual number of TNs will be constrained by physical capacity, real time, memory, and License limits.

Customer parameters

Table 20 "Customer parameters" (page 178) lists customer parameters and their maximum values.

Table 20
Customer parameters

Customer parameters	Maximum value	Comments
Tenants	512	
Dial Intercom Groups	2046	
Members per Dial Intercom Group	100	
Ringing Number Pickup groups	4095	Call Pickup Group 0 = no pickup group
Listed Directory Numbers (direct inward dialing only)	6	
DISA DNs	240	

Console and telephone parameters

Table 21 "Console and telephone related parameters" (page 179) lists console and telephone-related parameters and their maximum values.

Table 21
Console and telephone related parameters

Console/telephone parameters	Maximum value	Comments
Consoles per customer	63	
Lamp field arrays per customer	1	May be repeated once on another console.
Lamps per array (all numbers must be consecutive)	150	
Feature keys per attendant console: – M2250	 20	
Incoming call indicators per console	20	
Trunk group busy indicators per console: – M2250	 20	
Additional key/lamp strips: – console – telephones	 2 6	
Add on modules: – M3904 Key Expansion Module (KEM) – IP Phone 2002 KEM – IP Phone 2004 KEM	2 1 one-page KEM 2 one-page KEM or 1 two-page KEM	
Protect bcs block length	512	

Trunk and route parameters

Table 22 "Trunk and network-related parameters" (page 180) lists trunk and network-related parameters and their maximum values.

Table 22
Trunk and network-related parameters

Trunk/network parameters	Maximum value	Comments
Trunk routes per customer	512	
Members per trunk route	510	
RAN trunks per RAN route	10	
Trunk access restriction groups	32	
Locations in an ESN network	1000 or 16 000	1000 without ESN Location Code Expansion (LOCX) package 400; 16 000 with the LOCX package 400
Basic authorization codes	4096	
Length of basic authcode	14 digits	
Network authorization codes	20 000	ESN networks
Length of network authcode	7 digits	Fixed length defined per customer
NCOS: – CDP – BARS/NFCR – NARS/NSIG/AUTOVON	 3 7 15	
Route lists: – CDP – BARS – NARS	 32 128 256	
Route list entries	64	
NFCR trees	255	New Flexible Code Restriction
IDC trees	255	Incoming DID Digit Conversion
Virtual Trunk D-channels	64	

ACD feature parameters

Table 23 "ACD feature parameters" (page 181) lists ACD feature parameters and their maximum values.

Table 23
ACD feature parameters

ACD parameters	Maximum value	Comments
ACD DNs and CDNs per customer	-1000 (CP PII, CP PIV, CP PM)	The ACD-E package required, otherwise the limit is 240.
Agent positions per DN	-1200	Real-time and physical capacity constraints can limit this further.
Agent priorities	48	
Agent IDs per customer	9999	
Agents logged in at one time per system	9999	Real-time constraints can limit this further.
AST DNs per telephone	2	
Number of ACD-ADS customers	5	
Links per VASID	1	

Special feature parameters

Table 24 "Non-ACD feature parameters" (page 181) lists non-ACD feature parameters and their maximum values.

Table 24
Non-ACD feature parameters

Feature parameters	Maximum value	Comments
Speed call lists per system	8191	The number of speed call lists and the number of DNs per speed call list can be limited by the amount of available memory on the system (protected and unprotected data store).
Number of DNs in speed call list	1000	
Multiple appearances of the same directory number (MADN)	30*	Limited by watchdog timer. *See Steps in a hunting group.

Feature parameters	Maximum value	Comments
Steps in a hunting group	30*	Marketing objective, limited by watchdog timer. *In combination with MADN, each hunt step with more than 16 appearances is counted as two, so the maximum combination of MADN and hunt steps is 30 MADN and 15 hunt steps.
Number of Call Party Name Display names defined	Variable	Limited by the number of DNs defined and available space in the protected data store.
CPND length: – SL-1 protocol – ISDN protocol	 27 24	Software design limit. Display IE limitation (DMS switches have a display IE limit of 15).
AWU calls in 5 minutes	500	Marketing objective, constrained by ring generator.
Group Call Feature: –Groups per customer –Stations per group	 64 10	
BRI application: –Protocol parameter telephone groups per system –Terminal service profiles (per DSL) DSLs –LTIDs	 16 32 000 640 000	Software design limit; actual number is constrained by the number of TNs in the system. Each DSL occupies 2 TNs. Software design limit; each DSL can have a maximum of 20 LTIDs. The maximum number of LTIDs is limited by the number of DSLs, by memory, and by real time.

Hardware and capacity parameters

The software design limits are not typically the binding constraints. The number of items of a particular type is usually determined by a combination of loop and slot constraints (if the item requires loops) or by slot constraints alone.

Table 25 "Physical capacity/hardware-related parameters" (page 183) lists hardware and capacity parameters and their maximum values.

Table 25
Physical capacity/hardware-related parameters

Physical capacity/hardware parameters	Maximum value (loops)	Comments
MGCONF	MGC - 2 SSC - 4	Provide Conf, and MFS functionality for the cards in that MG 1000E
MGTDS	MGC - 2 SSC - 2	Provide TDS functionality for the cards in that MG 1000E.
Total service and terminal loops	256	Each superloop requires 4 loops.

Voice Gateway Media Cards

A Voice Gateway Media Card is any Media Card running the IP Line application.

In the CS 1000E, Voice Gateway Media Cards are used primarily for DSP connections between the TDM devices in an MG 1000E and IP circuits.

Voice Gateway Media Cards can be assigned to any slot other than slot 0. Each MG 1000E must be provisioned with enough Voice Gateway Media Cards to support the TDM devices in that MG 1000E.

Memory-related parameters

Table 26 "Memory-related parameters" (page 184) lists memory-related parameters and their maximum values.

Table 26
Memory-related parameters

Parameter	Values CS 1000E
Low-priority input buffers • (recommended default)	95 – 5000 (3500)
High-priority input buffers • (recommended default)	16 – 5000 (3500)
Input buffer size (words)	4
500-telephone, trunk and digital telephone output buffer • (recommended default)	16 –2048 (2000)
Message length (words)	4
D-channel input buffer size (bytes)	261
D-channel output buffer size (bytes)	266
TTY input buffer size (characters)	512
TTY output buffer size (characters)	2048
In a system with CallPilot, AML, and Symposium, add the number of CSQI and CSQO to the Call Register (CR) requirement obtained from feature impact calculations. The buffer estimates were based on relatively conservative scenarios, which should cover most practical applications in the field. However, most models deal with "average traffic". When traffic spikes occur, buffers can overflow. In these cases, raise the buffer size, depending on the availability of CRs. The maximum number of buffers allowed for CSQI and CSQO is 25% of NCR or 4095, whichever is less.	

Parameter	Values
	CS 1000E
Number of call registers • (recommended)	26 – 50 000 (25 000) 30 000 minimum call registers for systems with 22500 IP Phones.
Call registers assigned to AUX	26–255
Number of AML msg call registers	25 – the minimum of 25% of total call registers or 255 (default 25)
Call registers for CSL input queues (CSQI)	Maximum 25% of total call registers or 4095 (default 20)
Call registers for CSL/AML output queues (CSQO)	Maximum 25% of total call registers or 4095 (default 20)
Auxiliary input queue	20 – the minimum of 25% of total call registers or 255 (default 20)
In a system with CallPilot, AML, and Symposium, add the number of CSQI and CSQO to the Call Register (CR) requirement obtained from feature impact calculations. The buffer estimates were based on relatively conservative scenarios, which should cover most practical applications in the field. However, most models deal with "average traffic". When traffic spikes occur, buffers can overflow. In these cases, raise the buffer size, depending on the availability of CRs. The maximum number of buffers allowed for CSQI and CSQO is 25% of NCR or 4095, whichever is less.	

Parameter	Values CS 1000E
Auxiliary output queue	20 – the minimum of 25% of total call registers or 255 (default 20)
History file buffer length (characters)	0 – 65 535
In a system with CallPilot, AML, and Symposium, add the number of CSQI and CSQO to the Call Register (CR) requirement obtained from feature impact calculations. The buffer estimates were based on relatively conservative scenarios, which should cover most practical applications in the field. However, most models deal with "average traffic". When traffic spikes occur, buffers can overflow. In these cases, raise the buffer size, depending on the availability of CRs. The maximum number of buffers allowed for CSQI and CSQO is 25% of NCR or 4095, whichever is less.	

Buffer limits

The buffer limit is the maximum number of Call Registers (CR) that can be used for that particular function out of the total CR pool. If the designated limit is larger than needed and there are still spare CRs, the unused CRs will not be tied up by this specific function. Therefore, there is little penalty for overstating the buffer size limit, as long as the limit is within the number of CRs available to the system.

The values provided in Table 26 "Memory-related parameters" (page 184) indicate the relative requirements for various buffers. They are the minimum buffer size needed to cover most applications under the constraint of tight memory availability. When increasing buffer sizes, make the increases proportional to the values in Table 26 "Memory-related parameters" (page 184). This guideline applies in all cases except CSQI/CSQO, which is relatively independent of other buffers and can be increased without affecting others.

For example, with a CS 1000E Call Center (maximum 25 000 CRs) using many applications (such as CallPilot), it would be advisable to set the CSQI/CSQO to a high value (even up to the limit of 4095).

System capacities

Contents

This section contains information on the following topics:

"Introduction" (page 187)

"Memory size" (page 188)

"Mass storage" (page 190)

"Physical capacity" (page 190)

"CS 1000E network traffic" (page 193)

"Real-time capacity" (page 202)

"Signaling Server" (page 206)

"Software configuration capacities" (page 211)

"CS 1000E capacities" (page 212)

"Zone/IP Telephony Node Engineering" (page 213)

Introduction

This chapter describes the system’s primary capacity categories. For each category, this chapter:

- identifies the units that the capacity is measured
- details the primary physical and functional elements affecting the capacity
- describes actions that can be used to engineer the capacity

"Resource calculations" (page 215) provides the algorithms for engineering the system within the capacity limits. In some cases, applications such as Call Center require detailed engineering. These applications are discussed in "Application engineering" (page 265)

Memory size

Table 27 "CS 1000 Release 5.0 memory requirements" (page 188) shows the minimum amount of memory required for CS 1000 Release 5.0 software.

Table 27
CS 1000 Release 5.0 memory requirements

Processor	Flash memory required	DRAM memory required	Total memory
Call Server (CP PII)	N/A	256 MByte	256 MByte
Call Server (CP PIV)	N/A	512 MByte	512 MByte
Call Server (CP PM)	N/A	1 GByte	1 GByte
MG 1000E (SSC)	64 MByte	32 MByte	96 MByte
MG 1000E (MGC)	128 MByte	128 MByte	132 MByte
CP PIV and CP PM Call Servers can be upgraded to a maximum of 2 GByte DRAM memory			

Table 28 "Recommended maximum call register counts" (page 188) gives the maximum call register count recommended for CS 1000 Release 5.0 software, so that the system's memory requirements do not exceed the processor's memory capacity.

Table 28
Recommended maximum call register counts

System	Recommended call register count	Memory required (SL-1 words)	Memory required (MByte)
CS 1000E (CP PII)	25 000	6 057 000	23.174
CS 1000E (CP PIV and CP PM)	35 000	8 505 000	32.444
Call registers are 243 SL-1 words long. One SL-1 word is 4 bytes.			

Memory engineering

Current call processors for the CS 1000E are shipped with sufficient memory for the supported line sizes of the individual CPU types. Memory engineering is not required for most items.

Customer data is split between unprotected data store (UDS) and protected data store (PDS). Using LD 10 or LD 11 and looking at the memory usage, you can determine the amount of memory left on a system.

```
>ld 11
SL1000
MEM AVAIL: (U/P): 8064848 USED U P: 8925713 4998811 TOT: 21989372
```

The preceding example shows that there is 8,064,848 SL1 words (32,259,392 bytes) of memory left that can be used for either UDS or PDS. When the amount of available memory drops to be very low this will be shown as amount of UPS available and PDS available.

The preceding example also shows that currently 8,925,713 SL1 words (35,702,852 bytes) of UDS in use and 4,998,811 SL1 words (19,995,244 bytes) of PDS in use.

The major consumer of unprotected data store (UDS) is call register definitions. Therefore before increasing the number of call registers on a system, check that there is sufficient UDS available.

The major consumer of protected data store (PDS) is speed call lists. The overlay used to create speed call lists does the memory calculations (based on the number of lists, size of lists and DN sizes).

For definitions of large numbers of sets, it is recommended that you look that the available memory, create a single set and see how much memory was consumed. Then determine if there is sufficient memory left to create all of the desired sets.

Call register usage

Call register requirements on a system vary with usage and call patterns. In general you want at least 20% more call registers than sets, but this can vary with trunk usage or other features (ACD).

Assumptions:

- Call Register Traffic Factor (CRF) = 1.865
- The formula for calculating the recommended number of call registers depends on traffic load for the system.
- 28 CCS per ACD trunk
- Snacd = (Number of calls overflowed to all target ACD DNs × 2.25) – (Number of calls overflowed to local target ACD DNs × 1.8) (= 0 if the system is not a source node)
- Tnacd = 0.2 × Number of expected calls overflowed from source (= 0 if the system is not a target node)
- ISDN CCS = PRI CCS + BRI CCS
- ISDN penetration factor: p = ISDN CCS ÷ Total Voice Traffic

- ISDN factor: $(1 - p)2 + [4 \times (1 - p)] \times p + (3 \times p2)$

If Total Voice Traffic > 3000 CCS, then:

Recommended number of call registers = (CRF * 0.071 × Total Voice Traffic) + (0.33 × Number of ACD incoming trunks) + [(Snacd + Tnacd) × 0.03 × ISDN factor]

If Total Voice Traffic < 3000 CCS, then:

Recommended number of call registers = [(Number of system equipped ports – Number of ACD incoming trunks – Number of ACD agent telephones) × 0.94] + {(Number of ACD incoming trunks * 1.21) + [(Snacd + Tnacd) × 0.03]}

A general call register equation would be:

Recommended number of call registers = total ports + (total ports x trunking factor)

trunking factor = $(1 - p)^2 + [4 \times (1 - p)] \times p + (3 \times p^2)$

p (penetration factor) = trunking CCS ÷ Total Voice Traffic

Mass storage

The system processor program and data are loaded from hard disk or Compact Flash.

Software installation

Software, customer databases, and PEPS are delivered to the system using a Compact Flash card (RMD) inserted into the CP PIV or CP PM pack faceplate, or by using the floppy drive on CP PII faceplate. An installation process copies the software to the on-board Compact Flash (FMD). The software subsequently operates on this on-board Compact Flash (FMD).

Database backup

The faceplate Compact Flash (RMD) can also be used for customer database backups.

Physical capacity

A fully expanded CP PIV or CP PM CS 1000E system, with 50 MG 1000E each equipped with an MG 1000E Expander, provides 400 card slots (50 × 8) to support TDM devices and their required DSP resources. CS 1000E with CP PII is limited to 30 MG 1000Es.

A maximum of 256 loops are available to be used for gateway definitions, MGCONF and MGTDS definitions, and phantom or virtual loops for telephones and trunks. For more information on phantom and virtual loops, refer to the Global Software Licenses chapter in *Features and Services Fundamentals (NN43001-106)*.

Refer to "Assigning loops and card slots in the CS 1000E" (page 309) for information about loop and card slot usage and requirements for the Media Gateways in the CS 1000E.

Signaling and data links

The following signaling and data links are discussed in this section:

- "Physical links" (page 191)
- "Functional links" (page 191)

Physical links

There are three types of physical links to consider:

- "Serial Data Interface (SDI)" (page 191)
- "SDI through Terminal Server" (page 191)
- "Embedded Local Area Network (ELAN)" (page 191)

Serial Data Interface (SDI) The SDI is an asynchronous port, providing input access to the system from an OAM terminal and printing out maintenance messages to a TTY. The CP-PIV and CP-PM Call Processor cards have two ports, COM 1 and COM 2. COM 1 must be used for system installation and upgrades.

SDI through Terminal Server The SDI through the Terminal Server are asynchronous ports, providing input access to the system from an OAM terminal and printing out maintenance messages, traffic reports, and Call Detail Recording (CDR) records to a TTY. These asynchronous ports are programmed to connect automatically to PTY ports in the Call Server. For more information, refer to *Communication Server 1000E Installation and Commissioning (NN43041-310)*.

ATTENTION
A separate Terminal Server is not necessary when using a Media Gateway Controller card. Each MGC provides three remote SDI ports.

Embedded Local Area Network (ELAN) The system can communicate with a Host by Ethernet connection through a Network Interface Card (NIC). AML messages are embedded in the communication protocols, and they continue to interface with the system through CSQI and CSQO queues.

The data rate at the NIC port auto-negotiates up to 1000 MB full duplex.

Functional links

For each of the following functions, the type of link and resulting capacity are given.

Application Module Link (AML) AML is an Ethernet signaling link between the system and an Application Module (AM) connected to the ELAN subnet.

OAM The system uses a PTY port via a Terminal Server to connect to a terminal/computer (TTY) to receive maintenance commands or to print traffic reports, maintenance messages, or CDR records.

ISDN Signaling Link (ISL) An ISL provides common channel signaling for an ISDN application without PRI trunks. An analog trunk with modems at the originating switch and the terminating switch can be used as an ISL to transmit ISDN messages between these two remote systems. The interface for an ISL is an ESDI port. The maximum data rate for the link is 19.2 kbps.

D-Channel A PRI interface consists of 23 B-channels and 1 D-channel. The D-channel at 64 kbps rate is used for signaling. A D-channel communicates with the system through a DCHI card or a DHCI port on the D-channel handler. A D-channel on a BRI set is a 16 kbps link that is multiplexed to make a 54 kbps channel.

Property Management System Interface (PMSI) The PMSI lets the system interface directly to a customer-provided PMS through an SDI port in a Terminal Server. It is primarily used in Hotel/Motel environments to allow updates of the room status database either from the check-in counter or a guest room. The enhanced PMSI allows retransmission of output messages from the system to a PMS. The maximum baud rate for this asynchronous port is 9600.

Table 29 "I/O interface for applications" (page 192) summarizes the above functional links and interfaces and provides information required to calculate the number of I/O cards needed as an input to the card slot calculations.

Table 29
I/O interface for applications

Application	Type of link/ interface	Type of port	Sync or async
AML (associated telephone)	AML	ESDI	Sync
Symposium	ELAN	Ethernet	Sync
CallPilot	ELAN	Ethernet	Sync
CDR	RS232 C	PTY	Async
Host Enhanced Routing	AML	ESDI	Sync
Host Enhanced Voice Processing	CSL & AML	ESDI	Sync
An ESDI card has two ports; an SDI card has two ports; a DCHI card has one DCHI port and one SDI port.			

Application	Type of link/ interface	Type of port	Sync or async
ISL	Modem	ESDI	Sync
Interactive Voice Response	CSL	ESDI	Sync
Meridian 911	AML	ESDI	Sync
Property Management System Interface (PMSI)	PMSI Link	PTY	Async
NACD (PRI)	64 kB D-channel	DCHI	Sync
TTY (OAM)	RS232 C	PTY	Async
An ESDI card has two ports; an SDI card has two ports; a DCHI card has one DCHI port and one SDI port.			

CS 1000E network traffic

Traffic is a measure of the time a circuit is occupied. On the system, the circuit normally consists of a path from the telephone or trunk to the terminating telephone or trunk.

This section discusses the following traffic considerations:

- "Loops and superloops" (page 194)
- "Lines and trunks" (page 195)
- "Service loops and circuits" (page 196)
- "Voice Gateway Media Cards" (page 200)
- "Traffic capacity engineering algorithms" (page 200)

Terminology

Basic traffic terms used in this section are:

- ATTEMPT – any effort on the part of a traffic source to seize a circuit/channel/timeslot
- CALL – any actual engagement or seizure of a circuit or channel by two parties
- CALLING RATE – the number of calls per line per busy hour (Calls/Line)
- BUSY HOUR – the continuous 60-minute period of day having the highest traffic usage, usually beginning on the hour or half-hour
- HOLDING TIME – the length of time that a call engages a traffic path or channel
- TRAFFIC – the total occupied time of circuits or channels, generally expressed in CCS or Erlangs (CCS = a circuit occupied 100 seconds; Erlang = a circuit occupied one hour)

- BLOCKING – attempts not accepted by the system due to unavailability of the resource
- OFFERED traffic = CARRIED traffic + BLOCKED traffic
- Traffic load in CCS = Number of calls × AHT ÷ 100 (where AHT = average holding time)
- Network CCS = Total CCS handled by the switching network
  or
  CCS offered to the network by stations, trunks, attendants, Digitone Receivers, conference circuits, and special features

Loops and superloops

The number of loops and superloops on a CS 1000E is calculated form the number of lines, trunks, cards, and MG1000Es configured. The CS 1000E does not use the traffic requirements to determine loop usage.

Loop counting

- 1 Virtual Superloop has 1024 TNs (IP Phones, Vrtks)
- 2 MG 1000Es per Superloop (1000E PRI Gateway counts as 1 IPMG)
- MGC card
  - 1 loop per TDS definition (30 units per loop).
  - 1 loop per Conference definition (30 units per loop).
  - Can define up to 2 Conference loops and 2 TDS loops.
- SSC card
  - 1 loop per TDS definition (30 units per loop).
  - 1 loop per Conference definition (16 units per loop).
  - Can define up to 4 Conference loops and 2 TDS loops.
- 1 Phantom loop has 512 units. Used for M39xx Virtual Office.
- 1 Phantom loop has 1024 DECT users.
- 1024 i200x Virtual Office sets per Superloop.
- Every PRI definition requires 1 loop (23 channels TI, 30 channels E1).
- 1024 PCAs per Superloop.
- Limit of 64 Superloops (256 loops).
- Superloops = ROUNDUP(IP Phones + Vrtks) / 1024 + ROUNDUP (MGs / 2) + ROUNDUP(M39XX_vo / 512) + ROUNDUP(i200x_vo / 1024) + ROUNDUP(PCA / 1024) + ROUNDUP(DECT users / 1024)

- Loops = ROUNDUP(Conference Ports on MGC / 30) + ROUNDUP(Conference ports on SSC / 16) + TDS(#MGs) + PRI or DTI cards + (1000E PRI Gateways x 4)
- Total Loops - Loops + Superloops x 4
- Total Loops > 256 is an error, too many loops being used.
- Total Superloops = ROUNDUP(Total Loops / 4)

Superloop capacity

On a TDM based system (CS 1000M) each superloop is constrained by the number of talkslots and the number of CCS that the superloop can carry

The CS 1000E is an IP based system and does not have the same constraints. All Virtual superloops use "virtual talkslots" and are non-blocking (one virtual talkslot per virtual TN). This also removes the CCS per superloop constraint.

The MG1000E has a non-block TDM backplane (1 talksot per TDM unit). Call blocking can only occur here for other required resources (DTR, TDS, DSP, etc) which must all exist within the same MG1000E as the phone requiring the resource.

Loop capacity and MG 1000E TDM resources are subject to the Grade-of-Service (GoS) described under "Grade-of-Service" (page 200).

Lines and trunks

The relationship between lines and trunks is relevant for calculating loop requirements.

Voice over IP traffic

In the context of Voice over IP (VoIP) application, the lines include IP Phones and the trunks include IP Peer H.323 Virtual Trunks and SIP Virtual Trunks. The ratio of IP calls to the total line calls, and the ratio of H.323 and SIP Virtual Trunks calls to the total trunk calls, are required parameters. The split of TDM traffic to IP/Virtual Trunks (VT) becomes important, since resources such as Digital Signal Processor (DSP) in Media Cards and H.323 or SIP Virtual Trunks are affected by traffic distribution.

Figure 56 "CS 1000E system call types" (page 196) is a representation of the traffic flow for different types of calls. Each connection is denoted by a line. Only lines crossing the DSP line require a DSP port. For example, IP to IP connections in a CS 1000E system require no DSP and neither do IP to VT, but TDM to TDM do require DSP.

Figure 56
CS 1000E system call types

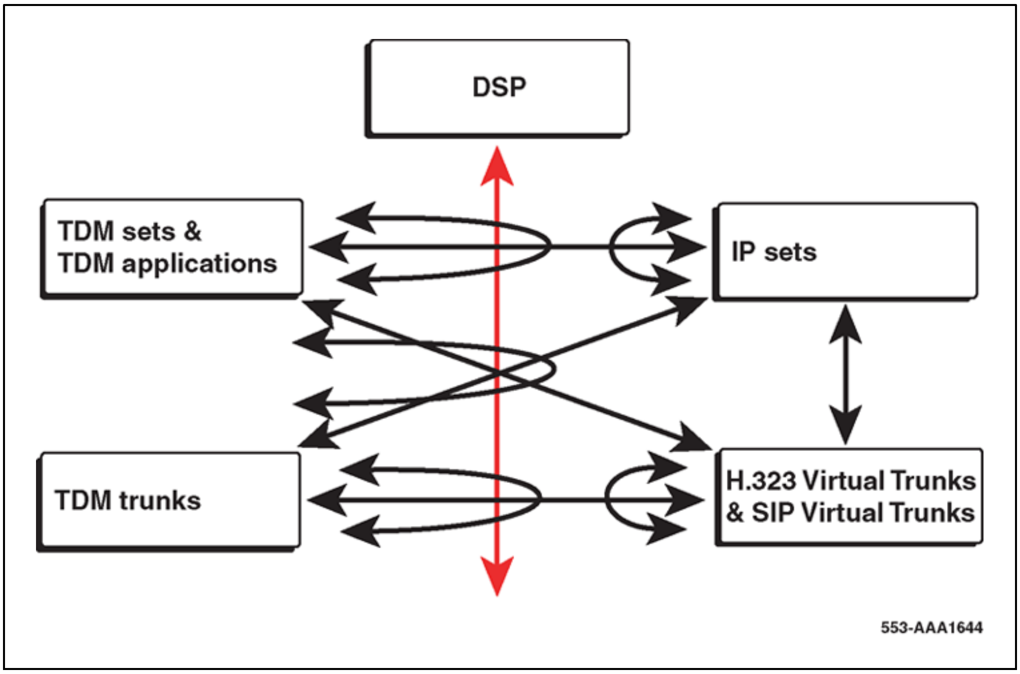


Table 30 "Connection type resources required" (page 196) lists the resources required for each type of connection.

Table 30
Connection type resources required

Connection Type	Resources
TDM to IP, IP to TDM	DSP
TDM to VT, VT to TDM	DSP and VT
IP to IP	no DSP
IP to VT or VT to IP	VT
TDM to TDM telephone or trunk calls	DSP no VT

Refer to "Resource calculations" (page 215) for the algorithms to calculate the required resources.

Service loops and circuits

Service circuits are required in call processing to provide specific functions to satisfy the requirements of a given application. Service circuits consume system resources, such as physical space, real time, memory, and so on.

In the CS 1000E, virtual tone and conference circuits (MGCONF and MGTDS) must be defined for use by each MG 1000E.

This section describes the traffic characteristics, calculation algorithms, and impact on other system resources of the following types of service circuits:

- "TDS" (page 197)
- "Conference" (page 197)
- "Broadcast circuits" (page 198)
- "DTR" (page 199)

TDS

The Tone and Digit Switch (TDS) loop provides dial tone, busy tone, overflow tone, ringing tone, audible ringback tone, DP or dual tone multifrequency (DTMF) outpulsing, and miscellaneous tones. All these tones are provided through the maximum 30 timeslots in the TDS loop.

A minimum of one TDS loop is required in each MG 1000E. The TDS circuits are provided by the MGC or SSC card. If additional TDS circuits are required in any MG 1000E, a second TDS loop can be configured in it. TDS circuits in an MG 1000E provide tones for TDM telephones or trunks in that MG 1000E only.

Conference

The MGC has a maximum of 2 conference loops, with 30 conference circuits for each conference loop, for a total of 60 conference circuits for each MGC-based MG 1000E. The maximum number of parties on a MG 1000E with MGC is 30.

The SSC has a maximum of 4 conference loops, with 16 conference circuits for each conference loop, for a total of 64 conference circuits for each SSC-based MG 1000E. The SSC has 32 conference circuits (2 loops) and another 32 conference circuits (2 loops) are on the dual-port IP daughterboard.

ATTENTION

If a CS 1000E system is equipped with a mix of SSC and MGC cards, and the MG 1000E with SSC has a conference loop configured, the maximum number of parties for any conference on the system is 6.

If a SSC single-port IP daughterboard is used, define no more than three conference loops for that MG 1000E. With a single-port daughterboard, there is no way to determine if a fourth conference loop exists, so a fourth conference loop causes failures for accessing conference circuits.

Conference circuits in the CS 1000E are a system resource

Broadcast circuits

The Nortel Integrated Recorded Announcer (Recorded Announcer) card provides either 8 or 16 ports to support Music, Recorded Announcement (RAN), and Automatic Wake Up. There is a maximum of 60 simultaneous connections to an individual card for broadcast within an MG 1000E. The use of controlled broadcast with Symposium and MGate cards has the same simultaneous connection limit as broadcast circuits. With special provisioning, the limit can be increased to 120 connections (see "Broadcast circuits" (page 317)).

Music

Music Broadcast requires any Music trunk and an external music source or a Recorded Announcer card. The Recorded Announcer has the capability to provide audio input for external music. A CON loop is not required for Music Broadcast.

Network Music

With the Network Music feature, a networked Central Audio Server is attached to the CS 1000E system to be used as the music source on demand to all parties on hold. With Network Music, the CS 1000E systems supports MOH features without a locally equipped music source for each node. Network Music feature provides music to every node in the system

The Central Audio Server is accessed over the network through H.323/SIP virtual trunks or TDM trunks. Virtual trunks or TDM trunks are connected to a network music trunk through an analog TIE trunk, the Network Music TIE trunk. Network Music is implemented with an XUT pack (NT8D14) and a network music agent. Broadcast music or conference music is set up so that multiple held parties can share the same music trunk.

To maximize the resource efficiency, the music is broadcast so that multiple parties can share the same music trunk. One music trunk can support a maximum of 64 listeners with broadcast music.

RAN

RAN trunks are located on eight-port trunk cards on PE shelves just like regular trunk circuits. They provide voice messages to waiting calls. RAN trunks are also needed to provide music to conference loops for music on hold.

Each RAN trunk is connected to one ACD call at a time, for the duration of the RAN message. Different RAN sources require different RAN trunk routes. If the first RAN is different from the second RAN, they need different RAN trunk routes. However, if the same message is to be used, the first RAN and second RAN can use the same route.

Use the following formula to calculate RAN traffic:

RAN CCS = Number of ACD calls using RAN × RAN HT ÷100

A RAN message typically runs from 20 seconds to 40 seconds. If the average for a specific application is not known, use a default of 30 seconds. After RAN CCS is obtained, estimate RAN trunk requirements from a Poisson P.01 table or a delay table (such as DTR table) matching the holding time of a RAN message.

DTR

A Digitone Receiver (DTR) serves features involving 2500 telephones or Digitone trunks. In CS 1000E systems, DTRs are not system-wide resources. They support only the telephones and trunks in the MG 1000E that they reside.

The MGC or SSC card provides 16 DTRs, or 8 DTRs and 4 Multifrequency Receivers (MFR). Additional DTRs can be provided by XDTR cards.

There are a number of features that require DTRs. General assumptions for DTR traffic calculations are:

- DTR traffic is inflated by 30% to cover unsuccessful dialing attempts.
- Call holding time used in intraoffice and outgoing call calculations is 135 seconds if actual values are unknown.
- DTR holding times are 6.2 and 14.1 seconds for intraoffice and outgoing calls, respectively.
- The number of incoming calls and outgoing calls are assumed to be equal if actual values are not specified.

The major DTR traffic sources and their calculation procedures are as follows:

Step	Action
1	Calculate intraoffice DTR traffic: Intraoffice = 100 × DTR station traffic (CCS) ÷ AHT × (R ÷ 2) (Recall that R is the intraoffice ratio.)
2	Calculate outgoing DTR traffic:

Outgoing = 100 × DTR station traffic (CCS) ÷ AHT × (1 - R ÷ 2)

3 Calculate direct inward dial (DID) DTR traffic:

DID calls = DID DTR trunk traffic (CCS) × 100 ÷ AHT

4 Calculate total DTR traffic:
Total = [(1.3 × 6.2 × intra) + (1.3 × 14.1 × outgoing calls) + (2.5 × DID calls)] ÷ 100

5 See "Digitone receiver load capacity 6 to 15 second holding time" (page 343) to determine the number of DTRs required. Note that a weighted average for holding times should be used.

—End—

Voice Gateway Media Cards

Since the Terminal Proxy Server (TPS) runs on the Signaling Server, Voice Gateway Media Cards do not carry traffic in the CS 1000E. They are used for DSP resources.

All the Voice Gateway Media Cards in a specific MG 1000E must be in the same zone, so that bandwidth management and codec selection can be performed properly.

Traffic capacity engineering algorithms

Traffic capacities of subsystems in the system are estimated based on statistical models that approximate the way a call is handled in that subsystem.

When inputs to the algorithm are lines, trunks, average holding time (AHT), and traffic load (CCS), the algorithms can be used to determine system size.

Alternatively, when the traffic capacity is known for a given configuration, the algorithms can be used to determine the traffic level allowed at the line and trunk level while meeting GoS requirements.

Grade-of-Service

In a broad sense, the Grade-of-Service (GoS) encompasses everything a telephone user perceives as the quality of services rendered. This includes:

- frequency of connection on first attempt
- speed of connection
- accuracy of connection
- average speed of answer by an operator

- quality of transmission

In the context of the system capacity engineering, the primary GoS measures are blocking probability and average delay.

Based on the EIA Subcommittee TR-41.1 Traffic Considerations for PBX Systems, the following GoS requirements must be met:

- Dial tone delay is not greater than 3 seconds for more than 1.5% of call originations.
- The probability of network blocking is 0.01 or less on line-to-line, line-to-trunk, or trunk-to-line connections.
- Blocking for ringing circuits is 0.001 or less.
- Post-dialing delay is less than 1.5 seconds on all calls.

Traffic models

Table 31 "Traffic models" (page 201) summarizes the traffic models that are used in various subsystem engineering procedures.

Table 31
Traffic models

Model	Assumptions	Service criteria	Applicability
Erlang B	Infinite sources (ratio of traffic sources to circuits > 5:1)	Blocked calls cleared (no queueing)	Loop, ringing circuit blocking
Erlang C	Infinite sources	Blocked calls delayed Infinite queue	Dial tone delay, I/O buffers, Digitone, RAN trunks
Poisson	Infinite sources	Blocked calls held for a fixed length	Incoming/outgoing trunks, Digitone, Call Registers, RAN trunks

Typically, the GoS for line-side traffic is based on Erlang B (or Erlang Loss formula) at P.01 GoS. When there is no resource available to process a call entering the system, the call is blocked out of the system. Therefore, the correct model to calculate the call's blocking probability is a "blocked call cleared" model, which is the basis of Erlang B.

When a call is already in the system and seeking a resource (trunk) to go out, the usual model to estimate trunk requirements is based on the Poisson formula. The reasons are:

- The Poisson model is more conservative than Erlang B (in that it projects a higher number of circuits to meet the same GoS). This reflects trunking requirements more accurately, since alternative routing (or routing tables) for outgoing trunk processing tends to increase loading on the trunk group.

- General telephony practice is to provide a better GoS for calls already using system resources (such as tones, digit dialing, and timeslots). Incomplete calls inefficiently waste partial resources. With more trunk circuits equipped, the probability of incomplete calls is lower.

Real-time capacity

Real-time capacity (load) refers to the ability of the Call Server to process instructions resulting from calls in accordance with service criteria.

Existing systems can use methods based on traffic data in order to determine Rated Call Capacity and current utilization levels. Refer to *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for a description of the TFS004 call capacity report and for information on interpreting TFS004 output.

If a new switch is being configured, equivalent basic calls must be calculated in order to estimate the processor loading of a proposed configuration.

Equivalent Basic Calls

An Equivalent Basic Call (EBC) is a measure of the real time required to process a basic call. A basic call is defined as a simple, unfeatured call between two IP Phones type 2004. The terminating telephone is allowed to ring once, then is answered, waits approximately two seconds, and hangs up. The originating telephone then hangs up as well.

When the capacity of a switch is stated in EBC, it is independent of such variables as configuration, feature mix, and usage patterns. It still varies from release to release, and between processors. However, since it is independent of other factors, it is a good way to compare the relative call processing capability of different machines running the same software release.

Table 32 "Real-time capacity (EBC) by system (with CS 1000 Release 5.0 software)" (page 202) gives the rated capacities of the Call Server processors in systems operating CS 1000 Release 5.0.

Table 32
Real-time capacity (EBC) by system (with CS 1000 Release 5.0 software)

System	Capacity
CS 1000E (CP PII processor)	210 000
CS 1000E (CP-PIV processor)	1 006 000
CS 1000E (CP-PM processor)	1 250 000

Feature impact

Every feature that is applied to a call increases the CP real time consumed by that call. These impacts can be measured and added incrementally to the cost of a basic call to determine the cost of a featured call. This is the basis of the algorithm used by EC to determine the rated capacity of a proposed switch configuration.

The incremental impact of a feature, expressed in EBC, is called the real-time factor for that feature. Real-time factors are computed by measuring the incremental real time for the feature in milliseconds, and dividing by the call service time of a basic call.

Each call is modeled as a basic call plus feature increments. For example, an incoming call from a DID trunk terminating on a digital telephone with incoming CDR is modeled as a basic call plus a real-time increment for incoming DID plus an increment for digital telephones plus an increment for incoming CDR.

A second factor is required to determine the overall impact of a feature on a switch. This is the penetration factor. The penetration factor is simply the proportion of calls in the system that invoke the feature.

The real-time impact, in EBC, of a feature on the system is computed as follows:

(Calls) × (penetration factor) × (real-time factor)

The sum of the impacts of all features, plus the number of calls, is the real-time load on the system, in EBC.

For penetration and real-time factors and for the detailed EBC calculations, refer to "System calls" (page 219) and "Real-time calculations" (page 222).

Call Server real-time calculations

The system EBC divided by the processor's rated capacity (see Table 32 "Real-time capacity (EBC) by system (with CS 1000 Release 5.0 software)" (page 202)) yields the fraction for processor utilization. This determines whether the proposed system can handle the load. If the projected real-time load is larger than the system capacity, a processor upgrade is needed.

Traffic peaking of 30% has been incorporated in the derivation of rated capacity. In other words, at 100% rated capacity, the absolute loading of the processor is 70%. Users should not adjust the rated capacity, but the loading percentage can reach 100% and the system can still function well. However, to preserve spare capacity for growth and extra traffic peaking, initial engineering of any site at full 100% loading is not recommended. A more typical initial load is about 85%.

If the configuration is an upgrade to an existing switch, in addition to calculating the new load as described above, users must also factor in CPU utilization data from a current traffic report TFS004. Users apply a formula to convert the existing processor usage to the equivalent loading on the new (and presumably faster) CPU.

Auxiliary processors

Interactions with auxiliary processors also have real-time impacts on the system CP depending on the number and length of messages exchanged. Several applications are described in "Application engineering" (page 265).

Real-time algorithm

As described above, calculating the real-time usage of a configuration requires information on the number of busy hour call attempts and the penetration factors of each feature.

Busy hour calls

If the switch is already running, the number of busy hour calls or call load can be determined from the traffic printout TFS004. The second field of this report (after the header) contains a peg count of CP Attempts. Examine a period of several days (a full week, if possible) to determine the maximum number of CP attempts experienced. This number varies with season, as well. The relevant number is the average of the highest ten values from the busiest four-week period of the year. An estimate is sufficient, based on current observations, if this data is not available.

If the switch is not accessible and call load is not known or estimated from external knowledge, call load can be computed. For this purpose, assumptions about the usage characteristics of telephones and trunks must be made. Refer to Table 37 "Major parameters for VoIP resource calculations" (page 216) for a description of the parameters that are required and default values, if applicable.

Telephones

As the primary traffic source to the system, telephones have a unique real-time impact on the system. For the major types listed below, the number of telephones of each type must be given, and the CCS and AHT must be estimated. In some cases it can be necessary to separate a single type into low-usage and high-usage categories. For example, a typical office environment with analog (500/2500-type) telephones can have a small call center with agents on analog (500/2500-type) telephones. A typical low-usage default value is 6 CCS. A typical high-usage default value is 28 CCS.

The principal types of telephones include:

- Analog: 500/2500-type, message waiting 500, message waiting 2500, and CLASS telephones

- Digital: M2000 series Meridian Modular Telephone, voice and/or data ports
- Consoles
- IP Phones 200x and 11xxE
- IP Softphone 2050

Trunks

Depending on the type of trunk and application involved, trunks can either be traffic sources, which generate calls to the system, or resources that satisfy traffic demands. Default trunk CCS in an office environment is 26 CCS. Call Center applications can require the default to be as high as 28 to 33 CCS.

Voice Analog:

- CO
- DID
- WATS
- FX
- CCSA
- TIE E&M
- TIE Loop Start

Digital:

- DTI: number given in terms of links, each provides 24 trunks under the North American standard
- PRI: number given in terms of links, each provides 23B+D under the North American standard
- European varieties of PRI, each provides 30B+D: VNS, DASS, DPNSS, QSIG, ETSI PRI DID

H.323 Virtual Trunk An IP Peer H.323 Virtual Trunk identified with a trunk route that is not associated with a physical hardware card.

SIP Virtual Trunk A Session Initiation Protocol (SIP) Virtual Trunk identified with a trunk route that is not associated with a physical hardware card.

Data

- Sync/Async CP
- Async Modem Pool

- Sync/Async Modem Pool
- Sync/Async Data
- Async Data Lines

RAN

The default value for AHT_{RAN} is 30 seconds.

Music

The default value for AHT_{MUSIC} is 60 seconds.

Signaling Server

The following software components operate on the Signaling Server:

- Terminal Proxy Server (TPS)
- H.323 Gateway (Virtual Trunk)
- SIP Gateway (Virtual Trunk)
- Network Routing Service (NRS)
- H.323 Gatekeeper
- Network Connection Service (NCS)
- CS 1000 Element Manager Web Server
- Application Server

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

A Signaling Server can also function as an application server for the Personal Directory, Callers List, and Redial List applications and Password administration. See "Application server for Personal Directory, Callers List, and Redial List" (page 210).

Table 33 "Elements in Signaling Server" (page 207) describes the function and engineering requirements of each element.

Table 33
Elements in Signaling Server

Element	Function and engineering requirements
Terminal Proxy Server (TPS)	• The TPS handles initial signaling exchanges between an IP Phone and the Signaling Server. • The TPS supports a maximum of 5000 IP Phones on each Signaling Server. • The TPS manages the firmware for the IP Phones that are registered to it. Accordingly, the TPS also manages the updating of the firmware for those IP Phones. • The redundancy of TPS is N+1. Therefore, one extra Signaling Server can be provided to cover TPS functions from N other servers.
H.323 Gateway (Virtual Trunk)	• The IP Peer H.323 Gateway trunk, or H.323 Virtual Trunk, provides the function of a trunk route without a physical presence in the hardware. The H.323 Gateway supports direct, end-to-end voice paths using Virtual Trunks. • The H.323 Signaling software (Virtual Trunk) provides the industry-standard H.323 signaling interface to H.323 Gateways. It supports both en bloc and overlap signaling. This software uses an H.323 Gatekeeper to resolve addressing for systems at different sites. • The H.323 Gateway supports up to 1200 H.323 Virtual Trunks per Signaling Server, assuming a combination of incoming and outgoing H.323 calls (see "Maximum number of SIP and H.323 Virtual Trunks" (page 209)). Beyond that, a second Signaling Server is required. 1 GByte of memory is required on Signaling Servers to support CS 1000 Release 5.0. If H.245 tunneling is not enabled, a maximum of 900 H.323 Virtual Trunks can be supported on a Signaling Server equipped with 1 GByte of memory. • The redundancy mode of the H.323 Gateway is 2 × N. Two H.323 Gateways handling the same route can provide redundancy for each other, but not for other routes.
The feasibility of combining the TPS, H.323 Gateway, SIP Gateway, and NRS on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

Element	Function and engineering requirements
SIP Gateway (Virtual Trunk)	• The SIP Gateway trunk, or SIP Virtual Trunk, provides a direct media path between users in the CS 1000E domain and users in the SIP domain. • The SIP trunking software functions as: – a SIP User Agent – a signaling gateway for all IP Phones • The SIP Gateway supports a maximum of 1800 SIP Virtual Trunks (see "Maximum number of SIP and H.323 Virtual Trunks" (page 209)). • The redundancy mode of the SIP Gateway is 2 × N. Two SIP Gateways handling the same route can provide redundancy for each other, but not for other routes.
Network Routing Service (NRS)	• The NRS has three components: – H.323 Gatekeeper – SIP Redirect Server – Network Connection Service (NCS) • The NRS must reside on the Leader Signaling Server. In a redundant configuration, the NRS is configured as Primary, alternate, or Failsafe (if required). • The NRS software limit for the combined total number of endpoints and routing entries is 20 000. The limit for the total number of endpoints is 5000 (up to 5000 SIP and up to 2000 H.323 endpoints). • The redundancy of the NRS is in a mode of 2 × N. An alternate NRS can serve only the NRS it is duplicating.
• H.323 Gatekeeper	• All systems in the network register to the H.323 Gatekeeper, which provides telephone number to IP address resolution. • The capacity of the H.323 Gatekeeper is limited by the endpoints it serves and the number of entries at each endpoint. • Potential hardware limits are the Signaling Server processing power and memory limits. • Since the Gatekeeper is a network resource, its capacity is a function of the network configuration and network traffic (IP calls). Some basic network information is required to engineer a Gatekeeper.
• SIP Redirect Server	• The SIP Redirect Server provides telephone number to IP address resolution. It uses a Gateway Location Service to match a fully qualified telephone number with a range of Directory Numbers (DN) and uses a SIP gateway to access that range of DNs.
• Network Connection Service (NCS)	• The NCS provides an interface to the TPS, enabling the TPS to query the NRS using the UNIStim protocol. The NCS is required to support the Media Gateway 1000B, Virtual Office, and Geographic Redundancy features.
The feasibility of combining the TPS, H.323 Gateway, SIP Gateway, and NRS on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

Element	Function and engineering requirements
CS 1000 Eleme nt Manager Web Server	• Has a negligible impact on capacity and can reside with any other element.
Application Server	• The Application Server for the Personal Directory, Callers List, and Redial List feature runs on the Signaling Server. • Only one database can exist in the network, and redundancy is not supported. • The database can co-exist with the other software applications on a Signaling Server. However, if there are more than 1000 users, Nortel recommends that the database be stored on a dedicated Signaling Server, (preferably a Follower). • The Application Server cannot be run on a Signaling Server at a branch office. • For more information on Personal Directory, Callers List, and Redial List, refer to *IP Line Fundamentals (NN43100-500)*.
The feasibility of combining the TPS, H.323 Gateway, SIP Gateway, and NRS on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

For more information about Signaling Servers, refer to *Signaling Server Installation and Commissioning (NN43001-312)*

Maximum number of SIP and H.323 Virtual Trunks

The maximum number of SIP and H.323 channels available on each Signaling Server depends on the number of available File Descriptors (FD) for Virtual Trunks. The maximum number of File Descriptors for Virtual Trunks is 1800.

- Each SIP call uses one FD.
- Each incoming H.323 call uses two FD.
- Each outgoing H.323 call uses one FD.

When no more File Descriptors are available (available FD = 0), new channels added on the Call Server cannot register on the Signaling Server.

Each Signaling Server supports up to 1800 Virtual Trunks. The maximum number of SIP and H.323 trunks depends on traffic patterns, both the split between SIP and H.323 calls and the split between incoming and outgoing

H.323 calls. Table 34 "Maximum number of Virtual Trunks, per Signaling Server" (page 210) gives examples of the maximum number of Virtual Trunks supported for different configurations.

Table 34
Maximum number of Virtual Trunks, per Signaling Server

SIP	H.323*			Total Virtual Trunks
	Incoming	Outgoing	Total H.323	
1800	0	0	0	1800
0	600	600	1200	1200
0	900	0	900	900
600	0	1200	1200	1800
600	300	600	900	1500
*Assumes H.245 tunneling is enabled.				

The formula to calculate the maximum number of Virtual Trunks is:

(Num_of_SIP × 1 FD) + (Num_of_Incoming_H323 × 2 FD) +
(Num_of_Outgoing_H323 × 1 FD) <= Max_Num_of_FDs
where Max_Num_of_FDs = 1800

Impact of H.245 tunneling

By default, H.245 tunneling is enabled. Unless there is a specific reason to disable tunneling, such as for maintenance, it should always be enabled. When tunneling is off, the handling capacity of the Signaling Server is reduced to a maximum of 900 H.323 Virtual Trunks.

Application server for Personal Directory, Callers List, and Redial List

The database for the Personal Directory, Callers List, and Redial List features for IP Phones must be located on one Signaling Server. The applications cannot be divided: all users in a system either have the combined Personal Directory, Callers List, and Redial List features or no feature at all. The Signaling Server can support a database for up to 9000 users.

- **Personal Directory**: Stores up to 100 entries per user of user names and DNs.
- **Callers List**: Stores up to 100 entries per user of caller ID information and most recent call time.
- **Redial List**: Stores up to 20 entries per user of dialed DNs and received Call Party Name Display with time and date.

The Signaling Server requires a minimum of 512 MByte of memory in order to support the Personal Directory, Callers List, and Redial List applications.

If the system size is relatively small, in terms of number of users as well as calling rates, one Signaling Server can serve both database and normal Signaling Server functions. With the Personal Directory, Callers List, and Redial List database co-resident with other applications (TPS, H.323/SIP Gateways, NRS, Element Manager), a Signaling Server with 512 MByte of memory can serve up to 1000 IP users and 382 Virtual Trunks. For larger systems, one additional Signaling Server, on top of the normal requirement for handling signaling traffic, is required for the Personal Directory, Callers List, and Redial List features.

The amount of memory required to support the Personal Directory, Callers List, and Redial List applications on the Signaling Server depends on the number of IP users and the configuration.Table 35 "Signaling Server memory requirements lists" (page 211) shows the memory requirements.

Table 35
Signaling Server memory requirements lists

Personal Directory, Callers List, and Redial List configuration	Number of IP users	Number of Virtual Trunks	Required memory
Co-resident with other applications	<= 1000	<= 382	1 GByte
Stand alone	1000 – 9000	> 382	1 GByte

ATTENTION

Signaling Servers require 1 GByte of memory to support CS 1000 Release 5.0.

ISP 1100 Signaling Servers will require a memory upgrade from 512 MByte to 1 GByte to be supported.

There is no redundancy for the Signaling Server dedicated to the Personal Directory, Callers List, and Redial List database. If that Signaling Server fails, the system loses those applications. However, the other Signaling Servers continue to function normally without the Personal Directory, Callers List, and Redial List features.

Software configuration capacities

The tables in "Design parameters" (page 177) provide maximum configuration capacities for applicable system and feature parameters. A system may not be able to simultaneously accommodate all of the maximum values listed because of system limitations on the real time, memory, or traffic capacity.

IP Telephony node maximums

The maximum number of Voice Gateway Media Cards per node is 30. When more than 30 Voice Gateway Media Cards are needed on a single CS 1000E system, use multiple nodes. The maximum number of Signaling Servers and Voice Gateway Media Cards combined within a node is 35.

CS 1000E capacities

Since IP telephony consumes less processing than TDM, the total number of telephones that a particular platform can support depends on the type of traffic as well as the physical capacity and applications of a specific configuration.

Table 36 "CS 1000E traffic capacities summary" (page 212) summarizes the capacities of CS 1000E systems. Values in each cell indicate the total number of telephones that can be supported in a particular configuration. These values are calculated from the point of view of call server processing capacity, not from the point of view of physical card slot capacity.

Values in each cell are exclusive, not additive.

Table 36
CS 1000E traffic capacities summary

Call server	Platform name	Total number of telephones			
		Pure TDM (no trunk ing)	IP telephones with access to PSTN	Pure IP (no access to PSTN)	Mixed TDM and IP telephones
CP PII	CS 1000E	3000	N/A	15 000	3000 TDM 10 000 IP
CP PIV or CP PM	CS 1000E	5000	22 500*	22 5000	5000* TDM 22 500* IP

Values in each column reflect the total telephones for a configuration. These are absolute limits for pure TDM and pure IP. For mixed TDM and IP, values are for typical configurations. Applications and calling patterns impact call server capacity. EC and NTPs are used to calculate practical values preconfiguration. Values beyond these limits must be engineered.

Requires using Signaling Servers for TPS.

Mixed configuration assumes 8-15% digital trunking to PSTN and no applications.

* Number of IP and TDM phones with PSTN access is dependant on the number of PRI/DTI/PRI2/DTI2 spans. This ties into loop consumption and must be calculated.

Zone/IP Telephony Node Engineering

For information on Zone/IP Telephony Node Engineering, refer to *Communication Server 1000M and Meridian 1 Large System Planning and Engineering (NN43021-220)*.

Resource calculations

Contents

This section contains information on the following topics:

"Introduction" (page 215)

"Resource calculation parameters" (page 216)

"System calls" (page 219)

"Real-time calculations" (page 222)

"DSP/Media Card calculations" (page 227)

"Virtual Trunk calculations" (page 232)

"Signaling Server algorithm" (page 235)

"Reducing imbalances (second round of algorithm calculations)" (page 251)

"Illustrative engineering example" (page 253)

Introduction

This chapter describes the algorithms implemented by the EC tool in order to calculate the resources required by the system.

In many cases, the calculations require user inputs that are the result of pre-engineering performed in accordance with the capacities and guidelines described in "System capacities" (page 187) and "Application engineering" (page 265).

When a proposed new system is equipped with more ports than the initial configuration actually uses, treat the two sets of input data like two separate configurations. Run each set of data through the algorithm and then compare results. For a viable solution, both sets of calculation results must be within the capacities of the proposed system.

Resource calculation parameters

Table 37 "Major parameters for VoIP resource calculations" (page 216) describes the major parameters used in the Voice over IP (VoIP) calculations. Some are user input and others are calculated.

Table 37
Major parameters for VoIP resource calculations

Parameter	Description	Equation	Default value
TDM telephone CCS (L_{TDM})	Sum of all digital and analog telephones and line-side T1/E1 ports, in CCS	(Number of digital telephones + Number of analog telephones + Number of line-side T1/E1 ports) × CCS per telephone	CCS per tele-phone: 5
IP telephone CCS (L_{IP}) (See Note 3 at end of table.)	Sum of all IP and IP ACD agent telephones, in CCS	[(Number of IP telephones – Number of IP ACD agents) × CCS per IP telephone] + (Number of IP agent telephones × CCS per agent)	CCS per tele-phone: 5
MDECT telephone CCS (L_{DECT})	Sum of all MDECT mobile telephones, in CCS	Number of MDECT telephones × CCS per telephone	CCS per tele-phone: 5
Total line CCS (L CCS)	Sum of all TDM, IP, and MDECT telephone CCS	TDM telephone CCS (L_{TDM}) + IP telephone CCS (L_{IP}) + MDECT telephone CCS (L_{DECT})	
TDM trunk CCS (T_{TDM})	Sum of all analog and digital trunks, in CCS	(Number of analog trunks + Number of digital trunks) × CCS per trunk	CCS per tele-phone: 26
Converged Desktop ratio (r_{DTP})	Of the total number of telephones, the portion that have the Converged Desktop feature	(Number of telephones with Converged Desktop) ÷ (Total number of telephones)	
Converged Desktop CCS (V_{DCCS}) (See Note 2 after the table.)	Converged Desktop CCS calculated as a percentage of total line CCS	Total line CCS (LCCS) × Converged Desktop ratio (r_{DTP})	
Microsoft Office Percentage (MOP)	Of the total number of telephones, the portion that have the Microsoft Converged Office feature.	(Number of telephones with Microsoft Converged Office) ÷(Total number of telephones)	

Parameter	Description	Equation	Default value
IP Security Enabled (IPSECP)	Calculation for IP Security	If any IP security features are enable set to 1, otherwise set to zero.	
H.323 Virtual Trunk CCS (HVT_{CCS})	Sum of all H.323 Virtual Trunks, in CCS	Number of H.323 Virtual Trunks (VT_{323}) × CCS per VT_{323}	
SIP Virtual Trunk CCS (SVT_{CCS})	Sum of all SIP Virtual Trunks, in CCS	Number of SIP Virtual Trunks (VT_{SIP}) × CCS per VT_{SIP}	
H.323 Virtual Trunk ratio (v_H)	Of total Virtual Trunk CCS, the portion that are H.323 Virtual Trunks	H.323 Virtual Trunk CCS (HVT_{CCS}) ÷ [H.323 Virtual Trunk CCS (HVTCCS) + SIP Virtual Trunk CCS (SVT_{CCS})]	
SIP Virtual Trunk ratio (v_S)	Of total Virtual Trunk CCS, the portion that are SIP Virtual Trunks	SIP Virtual Trunk CCS (SVT_{CCS}) ÷ [H.323 Virtual Trunk CCS (HVT_{CCS}) + SIP Virtual Trunk CCS (SVT_{CCS})]	
Virtual Trunk CCS (VT_{CCS})	Sum of H.323 Virtual Trunk CCS and SIP Virtual Trunk CCS	H.323 Virtual Trunk CCS (HVT_{CCS}) + SIP Virtual Trunk CCS (SVT_{CCS})	
Total trunk CCS (TT_{CCS})	Sum of all Virtual Trunk CCS and TDM trunk CCS	Virtual Trunk CCS (VT_{CCS}) + TDM trunk CCS (T_{TDM})	
Local CallPilot CCS (CP1) (See Note 4 after the table.)	CallPilot calls within the local node, calculated from number of local CallPilot ports	Local CallPilot ports × CCS per port ($CP1_{CCS}$)	
Network CallPilot CCS (CP2) (See Note 4 after the table.)	Network CallPilot calls to the local node, calculated from number of network CallPilot ports	Network CallPilot ports × CCS per port ($CP2_{CCS}$)	
IP ratio (P) (See Note 3 after the table.)	Of total line CCS, the portion that are from IP telephones	IP telephone CCS (LIP) ÷ Total line CCS (L_{CCS})	
Virtual Trunk ratio (V) (See Note 3 after the table.)	Of total trunk CCS, the portion that are from Virtual Trunk access ports	Virtual Trunk CCS (VT_{CCS}) ÷ Total trunk CCS (T_{TCCS})	
Total system CCS (T_{CCS})	Sum of all line and trunk CCS	Total line CCS (L_{CCS}) + Total trunk CCS (T_{TCCS})	

Parameter	Description	Equation	Default value
Intraoffice ratio (R_I)	Of the total number of calls, the portion that are telephone-to-telephone calls		0.30
Tandem ratio (R_T) (See Note 5 after the table.)	Of the total number of calls, the portion that are trunk-to-trunk calls		0.05
Incoming ratio (I)	Of the total number of calls, the portion that are trunk-to-telephone calls		0.40
Outgoing ratio (O)	Of the total number of calls, the portion that are telephone-to-trunk calls		
Average holding time (AHT_{XX})	Average holding time for different call types: Telephone-to-telephone (AHT_{SS}) Trunk-to-telephone (AHT_{TS}) — also used for ACD agents (AHT_{AGT}) Telephone-to-trunk (AHT_{ST})) Trunk-to-trunk (AHT_{TT})		60 sec 150 sec 150 sec 180 sec
Weighted average holding time (WAHT)		($R_I \times AHT_{SS}$) + (RT × AHT_{TT}) + (I × AHT_{TS}) + (O × AHT_{ST})	
Total calls (T_{CALL})	Total system calls per hour	$0.5 \times T_{CCS} \times 100 \div WAHT$	
Intraoffice calls (C_{SS})	Number of telephone-to-telephone calls	$R_I \times T_{CALL}$	
Tandem calls (C_{TT})	Number of trunk-to-trunk calls	$R_T \times T_{CALL}$	
Originating/outgoing calls (C_{ST})	Number of telephone-to-trunk calls	$O \times T_{CALL}$	
Terminating/incoming calls (C_{TS})	Number of trunk-to-telephone calls	$I \times T_{CALL}$	
DSP calls (C_{DSP})	Number of calls involving DSP		
Virtual Trunk calls (C_{VT})	Number of calls involving Virtual Trunks		
Conference loop ratio (r_{Con})	Ratio of conference loops to traffic loops	(Number of conference loops) ÷ (Total number of loops)	0.07

Note 1: In order to use the system traffic equations, all line-side T1/E1 and PRI trunks must be converted to number of ports. To convert T1 to ports: number of cards x 24. To convert E1 to ports: number of cards x 30.

Note 2: Converged Desktop traffic is part of the SIP Virtual Trunk traffic. The parameter value VDCCS must be less than the capacity of the number of SIP ports (VT_{SIP}).

Note 3: A site is considered to be a call center when the proportion of ACD agent telephones exceeds 15% of the total telephones in the system. For call centers, ACD agent calls are included in the calculations for Call Server usage. However, they are initially excluded from the calculations for DSP and Virtual Trunk resources. Once the DSP and Virtual Trunk resources have been calculated for non-ACD (reduced) traffic, the resources required to support the non-blocking ACD application (one DSP port for each ACD agent) are added back in to the results. The result is the total system DSP and Virtual Trunk requirements. The IP ratio (P) is modified for the non-ACD part of the calculation: P' = (L_{IP} without ACD)/(L_{TDM} without ACD + L_{IP} without ACD + L_{DECT}). The Virtual Trunk ratio (V) remains unchanged. The default traffic value for ACD agent telephones (TDM and IP) is 33 CCS per telephone.

Note 4: CallPilot message traffic is embedded in total line traffic. To calculate the real-time impact on the Call Server, CallPilot ports are converted to calls. Only CallPilot ports serving the local node (CP1) and handling network traffic (CP2) have a real-time impact on the Call Server.

Note 5: The tandem ratio should be kept at a relatively small number for a typical enterprise application, except when the switch serves as a tandem node in a network.

System calls

The total number of calls the system must be engineered to handle is given by:

$$\text{Total calls } (T_{CALL}) = 0.5 \times T_{CCS} \times 100 \div WAHT$$

where weighted average holding time (WAHT) is given by:

$$WAHT = (R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$$

and where AHT is the average holding time of a call in seconds.

The subscript indicates where the call initiated from and terminates on, with S = telephone and T = trunk. For example, AHT_{ST} denotes that the call initiated from a telephone and terminated on a trunk.

Traffic equations and penetration factors for CS 1000E

Total system calls comprise four different types of traffic:

1. Intraoffice calls (CSS) = Total calls (TCALL) × Intraoffice ratio (RI) (telephone-to-telephone) (Intraoffice calls (CSS) = Total calls (TCALL) × Intraoffice ratio (RI))
2. Tandem calls (CTT) = Total calls × Tandem ratio = TCALL × RT (trunk-to-trunk) (Tandem calls (CTT) = Total calls × Tandem ratio = TCALL × RT)
3. Originating/outgoing calls (CST) = Total calls × Outgoing ratio = TCALL × O (telephone-to-trunk) (Originating/outgoing calls (CST) = Total calls × Outgoing ratio = TCALL × O)
4. Terminating/incoming calls (CTS) = Total calls × Incoming ratio = TCALL × I (trunk-to-telephone) (Terminating/incoming calls (CTS) = Total calls × Incoming ratio = TCALL × I)

Intraoffice calls (telephone-to-telephone)

Intraoffice calls (C_{SS}) = Total calls (T_{CALL}) × Intraoffice ratio (R_I)

This parcel can be further broken down to three types:

1. Intraoffice IP to IP calls (C_{2IP})
   = $C_{SS} \times P^2$ (require no DSP, no VT)

   pf1 = $CSS \times P^2 \div T_{CALL} = R_I \times P^2$
   pf1 is the penetration factor for the intraoffice IP to IP calls
2. Intraoffice IP to TDM telephone calls (C_{1IP})
   = $C_{SS} \times 2 \times P \times (1 - P)$ (require DSP)

   pf2 = $C_{SS} \times 2 \times P \times (1 - P) \div T_{CALL} = 2 \times R_I \times P \times (1 - P)$
   pf2 is the penetration factor for the intraoffice IP to TDM telephone calls
3. Intraoffice TDM telephone to TDM telephone calls (C_{NoIP})
   = $C_{SS} \times (1 - P)^2$ (require two DSP, no VT)

   pf3 = $C_{SS} \times (1 - P)^2 \div T_{CALL} = R_I \times (1 - P)^2$
   pf3 is the penetration factor for the intraoffice TDM to TDM calls

Tandem calls (trunk-to-trunk)

Tandem calls (C_{TT}) = Total calls × Tandem ratio = $T_{CALL} \times R_T$

The tandem calls can be further broken down into:

- Tandem VT to TDM trunk calls (C_{T1VT})
  = 2 × Tandem VT calls × (1 – V)
  = $2 \times C_{TT} \times V \times (1 - V)$ (require DSP and VT)

  pf4 = $2 \times C_{TT} \times V \times (1 - V) \div T_{CALL} = 2 \times R_T \times V \times (1 - V)$
- Tandem TDM trunk to TDM trunk calls (C_{T2NoVT})

$= C_{TT} \times (1 - V)^2$ (require two DSP, no VT)

$pf5 = C_{TT} \times (1 - V)^2 \div T_{CALL} = R_T \times (1 - V)^2$

- Tandem VT (H.323) to VT (SIP) calls (C_{T2HS})
  $= C_{TT} \times V^2 \times v_H \times v_S \times 2 \times 2$ (require VT)

  $pf6 = 4 \times C_{TT} \times V^2 \times v_H \times v_S \div T_{CALL} = 4 \times R_T \times V^2 \times v_H \times v_S$

  where v_H is the fraction of H.323 trunks to total VTs, and vS is the fraction of SIP trunks to total VTs.

If there is only one type of VT (either v_H or $v_S = 0$), the connection is handled at the Network Routing Service and no calls are offered to the Call Server. In this case, pf6 = 0.

Originating/outgoing calls (telephone-to-trunk)

Originating/outgoing calls (C_{ST}) = Total calls × Outgoing ratio = $T_{CALL} \times O$

Originating/outgoing calls can be further broken down into:

1. IP to VT calls (C_{STIV})
   $= C_{ST} \times$ (fraction of IP calls) $\times V$
   $= C_{ST} \times P \times V$ (require VT)

   $pf7 = C_{ST} \times P \times V \div T_{CALL} = O \times P \times V$
2. IP to TDM trunk calls (C_{STID})
   $= C_{ST} \times$ (IP calls) $\times (1 - V)$
   $= C_{ST} \times P \times (1 - V)$ (require DSP)

   $pf8 = C_{ST} \times P \times (1 - V) \div T_{CALL} = O \times P \times (1 - V)$
3. TDM telephone to VT calls (C_{STDV})
   $= C_{ST} \times (1 -$ fraction of IP calls$) \times V$
   $= C_{ST} \times (1 - P) \times V$ (require DSP, VT)

   $pf9 = C_{ST} \times (1 - P) \times V \div T_{CALL} = O \times (1 - P) \times V$
4. TDM telephone to TDM trunk calls (C_{STDD})
   $= C_{ST} \times (1 -$ fraction of IP calls$) \times (1 - V)$
   $= C_{ST} \times (1 - P) \times (1 - V)$ (require two DSP, no VT)

   $pf10 = C_{ST} \times (1 - P) \times (1 - V) \div T_{CALL} = O \times (1 - P) \times (1 - V)$

Terminating/incoming calls (trunk-to-telephone)

Terminating/incoming calls (C_{TS}) = Total calls × Incoming ratio = $T_{CALL} \times I$

Terminating/incoming calls can be further broken down into:

1. VT to TDM telephone calls (C_{TSVD})
   $= C_{TS} \times V \times (1 -$ fraction of IP calls$)$
   $= C_{TS} \times V \times (1 - P)$ (require DSP, VT)

   $pf11 = C_{TS} \times V \times (1 - P) \div T_{CALL} = I \times V \times (1 - P)$

2. VT to IP telephone calls (C_{TSVI})
   $= C_{TS} \times V \times$ (fraction of IP calls)
   $= C_{TS} \times V \times P$ (require VT)

   $pf12 = C_{TS} \times V \times P \div T_{CALL} = I \times V \times P$

3. TDM trunk to IP telephone calls (C_{TSDI})
   $= C_{TS} \times (1 - V) \times$ (fraction of IP calls)
   $= C_{TS} \times (1 - V) \times P$ (require DSP)

   $pf13 = C_{TS} \times (1 - V) \times P \div T_{CALL} = I \times (1 - V) \times P$

4. TDM trunk to TDM telephone calls (C_{TSDD})
   $= C_{TS} \times (1 - V) \times (1 -$ fraction of IP calls)
   $= C_{TS} \times (1 - V) \times (1 - P)$ (require two DSP, no VT)

   $pf14 = C_{TS} \times (1 - V) \times (1 - P) \div T_{CALL} = I \times (1 - V) \times (1 - P)$

Resource use equations

The following equations, summing different types of traffic, are used to calculate the required TPS, DSP, and Virtual Trunk resources.

- Calls involving at least one IP Phone and therefore using TPS:

  $C_{IP} = C_{2IP} + C_{1IP} + C_{STIV} + C_{STID} + C_{TSVI} + C_{TSDI}$

- Calls that require DSP resources:

  (For CS 1000E)
  $C_{DSP} = C_{1IP} + (2 \times C_{NoIP}) + C_{T1VT} + (2 \times C_{T2NoVT}) + C_{STID} + C_{STDV} + (2 \times C_{STDD}) + C_{TSVD} + C_{TSDI} + (2 \times C_{TSDD})$

- Calls that require Virtual Trunk resources:

  $C_{VT} = C_{T1VT} + C_{T2HS} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI}$

  — Calls that require H.323 Virtual Trunks:

  $HC_{VT} = C_{VT} \times vH$

  — Calls that require SIP Virtual Trunks

  $SC_{VT} = C_{VT} \times v_S$

Real-time calculations

This section describes the following real-time calculations:

- "Call Server utilization" (page 225)
- "Application and feature EBCs" (page 225)
- "Call Server real time" (page 226)
- "CPU real-time conversion for upgrades" (page 226)

The real time required to process a basic IP Phone to IP Phone call is an Equivalent Basic Call (EBC), the unit used to measure other, more complicated feature calls. Every feature call can be converted to EBCs by using its real-time factor (RTF).

RTF = (Real time of a feature call in ms ÷ Real time of a basic call) – 1

There are a total of 14 major combinations of telephone and trunk types of calls in the system. The real-time factor of each call type is denoted by fi (i = 1 to 14). In addition, there are standard real-time factors for applications and features. Table 38 "Real-time factors" (page 223) provides the real-time factors.

Table 38
Real-time factors

Type of call	Real-time factor
Intraoffice calls:	
IP telephone to IP telephone (f_1)	0.00
IP telephone to TDM telephone (f_2)	2.00
TDM telephone to TDM telephone (f_3)	0.45
Tandem calls:	
Virtual Trunk to analog trunk (f_4)	2.09
Analog trunk to analog trunk (f_5)	2.09
H.323 Virtual Trunk to SIP Virtual Trunk (f_6)	1.93
Originating/outgoing calls:	
IP telephone to Virtual Trunk (f_7)	2.27
IP telephone to analog trunk (f_8)	2.42
TDM telephone to Virtual Trunk (f_9)	2.02
TDM telephone to analog trunk (f_{10})	1.27
Terminating/incoming calls:	
Virtual Trunk to TDM telephone (f_{11})	1.46
Virtual Trunk to IP telephone (f_{12})	1.60
Analog trunk to IP telephone (f_{13})	2.00
Analog trunk to TDM telephone (f_{14})	1.20
Application/feature calls:	
ACD agent without Symposium (f_{ACD})	0.13
Symposium (f_{SYM})	5.70
CallPilot (f_{CP})	1.70

Type of call	Real-time factor
Nortel Integrated Conference Bridge (f_{MICB})	1.59
Nortel Integrated Recorded Announcer (f_{MIRAN})	0.63
Nortel Integrated Call Assistant (f_{MICA})	0.57
Nortel Hospitality Integrated Voice Service (f_{MIVS})	0.57
Nortel Integrated Call Director (f_{MIPCD})	0.63
BRI ports (f_{BRI})	0.12
MDECT telephone (f_{DECT})	4.25
Intraoffice CDR (f_{ICDR})	0.44
Incoming CDR (f_{CCDR})	0.32
Outgoing CDR (f_{OCDR})	0.32
Tandem CDR ($_{fTAN}$)	0.44
CPND factor (f_{CPND})	0.20
Converged Desktop factor (f_{DTP})	2.33
Error term – minor feature overhead (f_{OVRH})	0.25
Microsoft Converged Office factor (f_{mo})	2.33
IP Security factor (f_{ipsec})	0.25

The real-time factor adjusts for the fact that a feature call generally requires more real time to process than a basic call. The impact on the system is a function of the frequency that the feature call appears during the busy hour. The penetration factor of a feature is the ratio of that type of feature call to the overall system calls. Refer to "Traffic equations and penetration factors for CS 1000E" (page 220) for the equations to calculate penetration factors for the 14 major call types.

The real-time factors and penetration factors are used to generate the real-time multiplier (RTM), which in turn is used to calculate the overall system EBC.

The real-time multiplier is given by:

$$\text{RTM} = 1 + \text{Error\_term} + \sum_{i} (\text{Real-time factor gi} \times \text{Penetration factor pfi})$$

The Error_term accounts for features such as call transfer, three-way conference, call-forward-no-answer, and others that are hard to single out to calculate real-time impact. The Error_term is usually assigned the value 0.25.

Call Server utilization

System real-time EBC = (Total system calls × Real-time multiplier) + Application and feature EBCs
= (T_{CALL} × RTM) + Application and feature EBCs

Application and feature EBCs

Table 39 "Application and feature EBCs" (page 225) lists the equations to calculate the EBC impacts of individual applications and features. The total application and feature EBC impact, which is included in the system real-time EBC calculation, is the sum of these application and feature EBCs.

Table 39
Application and feature EBCs

Type	Calculation
ACD	ACD agents without Symposium + ACD agents with Symposium where ACD agents without Symposium = (1 – % Symposium) × f_{ACD} × (Number of IP ACD agents + number of TDM agents) × CCS per agent × 100 ÷ AHT_{AGT} and ACD agents with Symposium is user input. (If unknown, assume all ACD agent calls are with Symposium.)
Symposium	% Symposium × f_{SYM} × (Number of IP ACD agents + number of TDM agents) × CCS per agent × 100 ÷ AHT_{AGT}
CallPilot	(Number of Local CallPilot ports + number of Network CallPilot ports) × CCS × 100 ÷ AHT_{CP} × f_{CP}
Internal CDR	C_{SS} × f_{ICDR}
Incoming CDR	C_{TS} × f_{CCDR}
Outgoing CDR	C_{ST} × f_{OCDR}
Tandem CDR	C_{TT} × f_{TCDR}
Integrated Conference Bridge	Number of Integrated Conference Bridge ports × CCS × 100 ÷ AHT_{MICB} × f_{MICB}
Integrated Recorded Announcer	Number of Integrated Recorded Announcer ports × CCS × 100 ÷ AHT_{MIRAN} × f_{MIRAN}
Integrated Call Director	Number of Integrated Call Director ports × CCS × 100 ÷ AHT_{MIPCD} × f_{MIPCD}
Integrated Call Announcer	Number of Integrated Call Announcer ports × CCS × 100 ÷ AHT_{MICA} × f_{MICA}
Hospitality Integrated Voice Services	Number of Hospitality Integrated Voice Services ports × CCS × 100 ÷ AHT_{MIVS} × f_{MIVS}

Type	Calculation
BRI	# BRI ports × CCS × 100 ÷ AHT_{BRI} × f_{BRI}
MDECT	L_{DECT} × 100 ÷ WAHT × f_{DECT}
CPND	(C_{1IP} + C_{NoIP} + C_{TSVD} + C_{TSDD}) × f_{CPND}
Converged Desktop (CD)	(C_{SS} × 0.1 + C_{TT} + C_{ST} + C_{TS}) × r_{DTP} × f_{DTP}
Microsoft Converged Office (MO)	(C_{ss} × 0.1 + C_{TT} + C_{ST}+ C_{TS}) × mop × fmo
IP Security (IPSEC)	(C_{ss} + C_{TT} + C_{ST} + C_{TS}) × P × ipsecp × fipsec

Call Server real time

Compare the system EBC with the system's CPU rated capacity to determine the processor utilization.

CPU utilization = System real-time EBC ÷ Rated capacity of processor (× 100 to get a percentage)

Refer to Table 32 "Real-time capacity (EBC) by system (with CS 1000 Release 5.0 software)" (page 202) for the rated capacities of CS 1000E processors.

CPU real-time conversion for upgrades

When upgrading an existing switch, CPU engineering must provide a certain level of spare capacity in order to ensure that the upgrade is able to handle both the existing site and the new additions. Real-time calculations must include the existing load as well as the new load.

The CPU utilization data from a current traffic report TFS004 provides the existing load. The existing load is then converted to the equivalent loading on the new (and presumably faster) CPU. The final loading on the new processor is the sum of the usual real-time calculations for the new load and the converted existing load. It must be less than or equal to 100% of the rated capacity for the new processor.

Use the following formula to convert the existing processor usage to the new processor equivalent:

CRTU = (RTU/100) × [1 + (SWRC ÷ 100)] × CPTU

where:

CRTU	=	CPU loading from the existing switch converted to an equivalent load on the new processor, in percent.
RTU	=	Current CPU usage, in percent (from the TFS004 report of the existing switch).

SWRC	=	Software release degradation factor, in percent. Since every new release is enhanced with new features and capabilities, the processing power of the existing CPU is degraded to some extent (typically 10-20%) by the newer release.
CPTU	=	Capacity ratio of the existing CPU to the new CPU. The ratio is always less than 1 (unless the same CPU is used, in which case it is equal to 1).

If CRTU > CPTU, telephone CRTU = CPTU.

Since the capacity ratio is the maximum load the old CPU can offer to the new one, the converted CPU load from the existing processor cannot be greater than the capacity ratio. Table 40 "Software release degradation factors (SWRC)" (page 227) lists the software release degradation factors for supported software upgrades.

Table 40
Software release degradation factors (SWRC)

From	Degradation factor (%) to CS 1000 Release 5.0
Rls 18	211
Rls 19	196
Rls 20B	143
Rls 21B	120
Rls 22	89
Rls 23	75
Rls 23C	71
Rls 24B	38
Rls 25B	27
SR2	25
SR3	23
SR4	18
SR4.5	8

DSP/Media Card calculations

DSP resources are provided by Media Cards. The total DSP/Media Card requirement is the sum of DSP requirements for various functions, which are calculated separately.

- "DSP ports for Conference" (page 229)

- "DSP ports for general traffic" (page 230)
- "DSP ports for major applications" (page 230)
- "Special ACD treatment for non-blocking access to DSP ports" (page 231)
- "Total DSP requirements" (page 232)
  - — "General configuration (ACD agent telephones less than 15 percent of total telephones)" (page 232)
  - — "Call center application (ACD agent telephones greater than 15 percent of total telephones)" (page 232)

For reasons explained in the ""System capacities" (page 187)" chapter (see "Traffic capacity engineering algorithms" (page 200)), the Erlang B model is used to calculate DSP port requirements.

For Media Card 32-port cards, the DSP port requirement must be calculated in increments of 32. Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228) provides Erlang B and Poisson values for P.01 Grade-of-Service (GoS) in 32-port increments. The DSP resource required to handle the offered traffic is the number of ports corresponding to the first Erlang B CCS capacity greater than the calculated traffic value. The Poisson values are used to calculate Virtual Trunk requirements (see "Virtual Trunk calculations" (page 232)).

Table 41
Erlang B and Poisson values, in 32-port increments

Erlang B with P.01 GoS		Poisson with P.01 GoS	
Number of DSP ports	**CCS**	**Number of Virtual Trunk access ports**	**CCS**
32	794	32	732
64	1822	64	1687
96	2891	96	2689
128	3982	128	3713
160	5083	160	4754
192	6192	192	5804

Because DSPs cannot be shared between MG 1000Es, the efficiency of the DSP ports on a Media Card is not as high as in a system-wide group. To calculate port and Media Card requirements, use the following (and round up to the next integer if the result is a fraction):

- 794 CCS per Media Card (32 ports)
- 1822 CCS per two Media Cards (64 ports)
- 2891 CCS per three Media Cards (96 ports)

For example, a calculated value of 2000 CCS requires 3 Media Cards to provide a P.01 GoS (2000 ÷ 1822 = 1.1 = 2 Media Cards + 1 Media Card to round up the fraction).

Refer to "Assigning loops and card slots in the CS 1000E" (page 309) for information on allocating Media Cards to MG 1000Es.

DSP ports for Conference

A DSP channel is required for each IP Phone joining a conference call. The more IP Phones in the system, the higher the demand for DSP channels to access the conference feature.

Additional DSP ports are required for TDM telephones in the CS 1000E. When a TDM telephone connects to a conference port on another MG 1000E, it uses one DSP port to reach the other MG 1000E and a second DSP port to connect to the conference port.

Applications are another source of demand for the conference feature. Conference usage for Integrated Conference Bridge is treated separately, as part of the calculations for application ports. For other applications, the default is two conference loops, with a total of 60 channels, per network group. If a particular application requires a different number of conference ports, use the specific number.

The equation to calculate the number of DSP ports the system requires for Conference is:

Equation 1
Number of DSP ports for Conference
= [(Total number of telephones) × rCon × 0.4]

where r_{Con} is the ratio of conference loops to traffic loops. The default value of r_{Con} is 0.07 because, for each network group, there are assumed to be 2 conference loops and 28 traffic loops (r_{Con} = 2÷ 28 = 0.07). The default value of r_{Con} can be changed if circumstances warrant.

DSP ports for digital trunks

Digital trunks typically carry heavy traffic (28 to 36 CCS) per port, therefore each digital trunk (PRI/PRI2/DTI/DTI2) requires a dedicated DSP port.

The minimum number of DSP ports required on the system equals the number of digital trunk ports on the system.

When digital trunk cards are installed in an MG1000E, that DSPs required to support that digital trunk card must be placed into the same MG1000E.

DSP ports for general traffic

There are three steps to calculate the number of DSP ports required for general traffic:

1. Calculate the number of calls that require DSP resources.

   DSP calls (C_{DSP}) = Intraoffice IP-TDM telephone calls (C_{1IP}) + [2 × Intraoffice TDM telephone-TDM telephone calls (C_{NoIP})] + Tandem VT-TDM trunk calls (C_{T1VT}) + [2 × Tandem TDM trunk-TDM trunk calls (C_{T2NoVT})] + IP-TDM trunk calls (C_{STID}) + TDM telephone-VT calls (C_{STDV}) + [2 × TDM telephone-TDM trunk calls (C_{STDD})] VT-TDM telephone calls (C_{TSVD}) + TDM-IP telephone calls (C_{TSDI}) + [2 × TDM trunk-TDM telephone calls (C_{TSDD})]

   $= C_{1IP} + (2 \times C_{NoIP}) + C_{T1VT} + (2 \times C_{T2NoVT}) + C_{STID} + C_{STDV} + (2 \times C_{STDD}) + C_{TSVD} + C_{TSDI} + (2 \times C_{TSDD})$

   For sites where the proportion of ACD agent telephones is less than 15% of the total telephones in the system, CDSP includes all general traffic seeking DSP service.

   Sites where the proportion of ACD agent telephones exceeds 15% of the total telephones in the system are considered to be call centers. For call centers, CDSP is a reduced total that excludes ACD CCS. See "Special ACD treatment for non-blocking access to DSP ports" (page 231) and Note 3.

2. Convert DSP calls to CCS.

   DSP CCS = $C_{DSP} \times WAHT \div 100$

3. Using the Erlang B table for P.01 GoS (see Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228)), find the corresponding number of DSP ports required.

   Equation 2

   Number of DSP ports for general traffic = Required number of ports for DSP CCS from Erlang B table

DSP ports for major applications

Provide one DSP port for each application port:

- Integrated Recorded Announcer ports
- Integrated Conference Bridge ports

- Integrated Call Director ports
- Integrated Call Assistant ports
- Hospitality Integrated Voice Service ports
- BRI (SILC ports)
- CallPilot ports
- Agent Greeting ports

Equation 3

Number of DSP ports for applications = DSP for Integrated Recorded Announcer + DSP for Integrated Conference Bridge + ... + DSP for Agent Greeting ports

Special ACD treatment for non-blocking access to DSP ports

The following section applies for call centers, which are defined as sites where the number of ACD agent telephones exceeds 15% of the total telephones in the CS 1000E.

Since both Erlang B and Poisson models assume a high ratio of traffic sources to circuits, using the standard estimate of 36 CCS per agent to calculate DSP requirements for a specified GoS tends to result in over-provisioning. For this reason, rather use the fixed rule of one DSP port for each ACD agent telephone requiring a DSP resource, in order to provide non-blocking access between an ACD agent telephone and a DSP. ACD agent telephones require DSP resources only when calls are coming from TDM trunks to IP agent telephones or from Virtual Trunks to TDM agent telephones.

The ACD agent telephones must be located in the same MG 1000E as the Media Cards providing the DSP resources for the telephones.

Assuming that a group of Media Cards can be reserved for the exclusive use of ACD agents, recalculate the number of DSP ports required for general traffic excluding ACD agent CCS, and then add in DSP ports required for the ACD agent telephones. The steps are as follows:

1. Calculate system CCS excluding ACD agents. Since system CCS is two-way traffic, the traffic associated with both incoming and outgoing trunks terminating on ACD agents must be removed:

   Reduced system CCS = Total system CCS (T_{CCS}) – [2 × (Number of ACD agent telephones) × CCS/agent]

2. Recalculate the intraoffice ratio (RI), IP ratio (P), Virtual Trunk ratio (V), and other ratios to reflect the new distribution of call types without ACD traffic. (See Table 37 "Major parameters for VoIP resource calculations" (page 216) for the equations to calculate the ratios. See also Note 3.)

3. Use the reduced system CCS and new ratios to calculate calls requiring DSP and Virtual Trunk resources. (See "Traffic equations and penetration factors for CS 1000E" (page 220) for the detailed calculations for the different call types.)
4. Convert DSP calls to CCS.

   DSP CCS = C_{DSP} × WAHT ÷ 100
5. Using the Erlang B table for P.01 GoS (see Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228)), find the corresponding number of DSP ports required (for general traffic without ACD agents).

   Equation 2a

   Number of DSP ports for general traffic = Required number of ports for DSP CCS from Erlang B table
6. Calculate the DSP requirement for ACD agent telephones. A DSP port is needed when calls are coming from TDM trunks (ratio 1 – V) to IP agent telephones or from Virtual Trunks (ratio V) to TDM agent telephones.

   Equation 4

   Number of DSP ports = (Number of IP ACD agent telephones) × (1 – V) + (Number of TDM ACD agent telephones) × V
7. In the CS 1000E, additional DSP ports are required for calls from analog trunks to TDM ACD agent telephones in another MG 1000E.

   Equation 4e

   Additional DSP ports = 2 × (Number of TDM ACD agent telephones) × (1 – V)

Total DSP requirements

General configuration (ACD agent telephones less than 15 percent of total telephones)

Total number of DSP ports = Equation 1 + Equation 2 + Equation 3

Call center application (ACD agent telephones greater than 15 percent of total telephones)

Total number of DSP ports = Equation 1 + Equation 2a + Equation 3 + Equation 4 + Equation 4e

Virtual Trunk calculations

For reasons explained in the ""System capacities" (page 187)" chapter (see "Traffic capacity engineering algorithms" (page 200)), the Poisson model is used to calculate trunk requirements.

Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228) provides Poisson values for P.01 GoS in 32-port increments. The Virtual Trunk resource required to handle the offered traffic is the number of access ports corresponding to the first Poisson CCS capacity greater than the calculated traffic value.

To obtain the exact number of access ports required, use the following formula. Round up to the next integer if the result is a fraction.

Number of access ports = (Calculated CCS) ÷ (CCS from Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228)) × (Number of access ports for table CCS)

Perform the following steps to calculate the number of access ports required:

1. Estimate the Virtual Trunk requirement by adding together all the calls that require the service of access ports.

   Virtual Trunk calls (C_{VT}) = Tandem VT-TDM trunk calls (C_{T1VT}) + IP-VT calls (C_{STIV}) + TDM telephone-VT calls (C_{STDV}) + VT-TDM telephone calls (C_{TSVD}) + VT-IP telephone calls (C_{TSVI}) + H.323-SIP VT calls (C_{T2HS})

   = $C_{T1VT} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} + C_{T2HS}$

   For sites where the proportion of ACD agent telephones is less than 15% of the total telephones in the system, C_{VT} includes all general traffic seeking an access port.

   Sites where the proportion of ACD agent telephones exceeds 15% of the total telephones in the system are considered to be call centers. For call centers, CVT is a reduced total that excludes ACD CCS. See "Special ACD treatment for non-blocking access to DSP ports" (page 231) and Note 3.

2. Convert Virtual Trunk calls to CCS.

   Virtual Trunk CCS (VT_{CCS}) = C_{VT} × WAHT ÷ 100

3. For call centers, since the calculated Virtual Trunk calls exclude ACD traffic, restore ACD traffic so that the final number of Virtual Trunks is sufficient to handle both general and ACD traffic.

   Final Virtual Trunk CCS = (Calculated VTCCS without ACD) + [(Number of IP ACD agent telephones) + (Number of TDM ACD agent telephones)] × V × (CCS per ACD agent) ÷ (CCS per trunk)
   where:
   default CCS per ACD agent = 33 CCS
   default CCS per trunk = 28 CCS

   The expanded Virtual Trunk CCS is inflated by the ratio of 33/28 to reflect the fact that more Virtual Trunks are needed to carry each agent

CCS. This is because the traffic levels engineered for ACD agents and Virtual Trunks are different.

4. Use the SIP and H.323 ratios to determine how the Virtual Trunk access ports are to be allocated to the two groups.

   SIP Virtual Trunk CCS (SVT_{CCS}) = $VT_{CCS} \times v_S$
   H.323 Virtual Trunk CCS (HVT_{CCS}) = $VT_{CCS} \times v_H$

5. Using the Poisson table for P.01 GoS (see Table 41 "Erlang B and Poisson values, in 32-port increments" (page 228) or "Trunk traffic Erlang B with P.01 Grade-of-Service" (page 336)), find the corresponding number of SIP and H.323 access ports required.

   Although a Virtual Trunk does not need the physical presence of a superloop, it does utilize a logical superloop. A superloop of 128 timeslots can support 1024 Virtual Trunk channels.

Reducing Virtual Trunk imbalances

The final value for calculated Virtual Trunks and its split into SIP and H.323 can be different from initial user input. If the gap between user input and the calculated result is less than 20%, use either number (although the larger number is preferred). If the gap is bigger, the configuration is not balanced. It can be necessary to re-enter input data, including other input parameters, and fine tune the configuration in order to narrow the gap. See "Reducing imbalances (second round of algorithm calculations)" (page 251).

A discrepancy between calculated and input Virtual Trunks is significant because system resources such as DSP ports and Virtual Trunk licenses depend on the accuracy of the traffic split. Imbalanced Virtual Trunk traffic renders the resulting equipment recommendation unreliable.

For example, if the calculated number of Virtual Trunks is 80 but the original input value was 60, and the user decides to use the original input value of 60 to calculate bandwidth and Signaling Server requirements, the resulting system will likely provide service inferior to the normal expected P.01 GoS. On the other hand, if the user input was 80 and the calculated result is 60, it is up to the user to choose the number to use for further calculations for necessary resources, such as the LAN/WAN bandwidth requirement. Unless the configuration is constrained in some way, the larger of the two values (input number or calculated number) is always preferred.

Bandwidth requirement for access ports

The LAN/WAN bandwidth requirement is based directly on traffic. Therefore, it does not depend on the traffic model used nor on the number of Virtual Trunks (either input or calculated) used for other calculations.

Convert Virtual Trunk calls to erlangs:

VT erlangs = VTCCS ÷ 36

Look up the VT erlangs number in a bandwidth table to find the corresponding bandwidth required to carry the Virtual Trunk traffic to other H.323 endpoints. Refer to Converging the Data Network with VoIP Fundamentals (NN43001-260) for the bandwidth table and for more information about calculating LAN/WAN bandwidth requirements.

Signaling Server algorithm

The Signaling Server algorithm in the EC tool determines the number of Signaling Servers required for a given configuration. The algorithm allows a change in constants for Signaling Server platform or Signaling Server application software releases.

The software components that operate on the Signaling Server are the Network Routing Service (NRS), Terminal Proxy Server (TPS), IP Peer Gateways (H.323 and SIP), and Element Manager. Traffic and user requirements determine whether the software components share a Signaling Server or are served by stand-alone Signaling Servers.

For the applications, there are performance factors and software limit factors. The performance factors are determined through capacity analysis. The software limit factors are defined by the application. Element Manager can collocate with any of the other applications with negligible impact.

In order to calculate the number of Signaling Servers required to support a particular configuration, the algorithm first calculates the amount of Signaling Server resources required by each application, taking redundancy requirements into consideration. The calculation for each application is performed separately. Once the individual requirements are determined, the algorithm proceeds to evaluate sharing options. Then the results are summed to determine the total Signaling Server requirement.

In most cases, the individual calculations divide the configuration's requirement for an applicable parameter (endpoint, call, telephone, trunk) into the system limit for that parameter. The particular application's Signaling Server requirement is determined by the parameter with the largest proportional resource requirement, adjusted for redundancy.

Table 42 "Signaling Server algorithm constant and variable definitions" (page 236) defines the variables used in the calculations.

Table 42
Signaling Server algorithm constant and variable definitions

Algorithm term	Description	Value	Notes
NRS[type_index]	Network Routing Service (NRS) required	enter (Variable to be entered into the formula)	Yes or No.
NRS_Linux[type_index]	Network Routing Service (NRS) Linux required	enter (Variable to be entered into the formula)	1 if SIP Proxy; otherwise 0
NRA[type_index]	Network Routing Service (NRS) alternate required	enter (Variable to be entered into the formula)	Yes or No.
NRC[type_index]	NRS calls per hour	enter (Variable to be entered into the formula)	Two components (one local, one network): $NRC = NRC_0 + NRC_{NET}$
NRC_{HL}[type_index]	NRS calls per hour	100 000 (Constant in the formulas.)	Hardware limit for Signaling Server.
NRC_{SL}[type_index]	NRS calls per hour	10 000 (Constant in the formulas.)	Software limit for Signaling Server.
NRD[type_index]	NRS CDP + UDP entries	enter (Variable to be entered into the formula)	
NRD_1[type_index]	NRS CDP + UDP entries limit	20 000 (Constant that updates with system software releases.)	Software limit.
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Algorithm term	Description	Value	Notes
NRD_{HL}[type_index]	NRS product of endpoint and CDP/UDP entries	20 000 (Constant in the formulas.)	Hardware limit.
NRD_{SL}[type_index]	NRS product of endpoint and CDP/UDP entries	1000 (Constant in the formulas.)	Shared limit.
NRE[type_index]	NRS endpoints	enter (Variable to be entered into the formula)	(= 0 if NRS, which is a network-wide resource, is not provisioned in this node)
NRE_1[type_index]	NRS endpoints limit	5000 (Constant that updates with system software releases.)	Software limit.
NRE_{SL}[type_index]	NRS endpoints limit	100 (Constant in the formulas.)	Shared limit.
NRP[type_index]	NRS product of endpoint and CDP/UDP entries	- (Constant that updates the platform changes.)	Interim calculation.
NRP_{SL}[type_index]	NRS product of endpoint and CDP/UDP entries	20 000 (Constant that updates with system software releases.)	Software limit.
GSA[type_index]	SIP Gateway alternate required	enter (Variable to be entered into the formula)	Yes or No.
GWA[type_index]	H.323 Gateway alternate required	enter (Variable to be entered into the formula)	Yes or No.
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Algorithm term	Description	Value	Notes
C_{IP}[type_index]	Internet phone calls per hour	enter/ derived (Variable to be entered into the formula)	Busy Hour calls from all IP Phones.
IPC_{HL}[type_index]	Internet phone calls per hour limit	40 000 (Constant that updates with system software releases.)	Hardware limit.
IPL[type_index]	Internet phones	enter (Variable to be entered into the formula)	
IPL_{DB}[type_index]	IP Phone limit with Personal Directory/Callers List/Redial List database	1000 (Constant in the formulas.)	IP Phone limit per Signaling Server reduced due to Personal Directory/Callers List/Redial List database.
IPL_{SL}[type_index]	IP Phones limit	5000 (Constant that updates with system software releases.)	Software limit.
SSNR[type_index]	NRS Signaling Server calculation	calc (Calculated result)	Real number requirement (e.g., 1.5) (= 0 if NRS is not provisioned in this node)
SSNW[type_index]	NRS Signaling Server requirements	calc (Calculated result)	Whole number requirement including alternate.
SSNW_Linux[type_index]	NRS Signaling Server requirements	calc (Calculated result)	Whole number requirement including alternate.
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Algorithm term	Description	Value	Notes
SSLinux[type_index]	NRS Linux Signaling Server requirements	calc (Calculated result)	Whole number requirement including alternate.
SPSA[type_index]	NRS Linux alternate required	enter (Variable to be entered into the formula)	Yes or No
SSHR[type_index]	H.323 Gateway Signaling Server calculation	calc (Calculated result)	Real number requirement (e.g.,1.5).
SSHW[type_index]	H.323 Gateway Signaling Server requirements	calc (Calculated result)	Whole number requirement including alternate.
SST[type_index]	Total count of Signaling Servers required	calc (Calculated result)	
SSTR[type_index]	TPS Signaling Server calculation	calc (Calculated result)	Real number requirement (e.g.,1.5).
SSTW[type_index]	TPS Signaling Server requirements	calc (Calculated result)	Whole number requirement including alternate.
TPSA[type_index]	TPS N+1 redundancy required	enter (Variable to be entered into the formula)	Yes or No.
$HVTC[type\_index]_{HL}$	H.323 Gateway calls per hour limit	18 000 (Constant that updates with system software releases.)	Hardware limit. $HVTC_{HL} = v_H \times C_{VT}$
$SVTC[type\_index]_{HL}$	SIP Gateway calls per hour limit	27 000 (Constant that updates with system software releases.)	Hardware limit. $SVTC_{HL} = v_S \times CVT$
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Algorithm term	Description	Value	Notes
VT_{SIP}[type_index]	SIP Gateway access ports per Signaling Server	1800 (Constant that updates with system software releases.)	CPU limit.
VT_{323}[type_index]	H.323 Gateway access ports per Signaling Server	1200 (Constant that updates with system software releases.)	CPU limit.
VT_{SL}[type_index]	Virtual Trunks shared limit	382 (Constant that updates with system software releases)	Virtual Trunks co-res shared limit.
TR87[type_index]	The aggregate number of SIP CTI/TR87 required based upon the MCS and LCS calculated.	enter	
$TR87_{CL}$[type_index]	SIP CTI/TR87 clients	5000 (Constant that updates with system software releases)	CPU limit.
$TR87_{SL}$[type_index]	SIP CTI/TR87 calculation	100 (Constant in the formula)	SIP CTI/TR87 co-res limit
TR87A[type_index]	SIP CTI/TR87 redundancy required	enter (Variable to be entered into the formula)	Yes or No
SSTR87W[type_index]	SIP CTI/TR87 Signaling Server requirements	calc (Calculated result)	Whole number required including alternate.
SSTR87[type_index]	SIP CTI/TR87 calculation	calc (Calculated result)	Real number requirement.
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Algorithm term	Description	Value	Notes
SS_Co_Res	Signaling Server co-residency sum	calc (Calculated result)	Sum of co-res ratio
Co_Res_VT	Virtual Trunk co-residency sum	calc (Calculated result)	1 if VT can co-res; otherwise 0
Co_Res_PD	PD co-res with TPS and GW	calc (Calculated result)	1 if PD can co-res; otherwise 0
Signaling Server hardware can be ISP1100, CP PM, IBM, or HP servers. For the calculations, each variable is indexed by the Signaling Server type. type_index = ISP or CPPM or IBM or HP			

Signaling Server calculations

All the Signaling Server software components can function either on shared or on stand-alone Signaling Servers. System traffic and user requirements (for alternate, redundant, or dedicated Signaling Servers) determine how many Signaling Servers are required. The Signaling Server algorithm takes account of all these requirements by performing the following calculations in sequence:

1. Signaling Server for Personal Directory/Callers List/Redial List database (SSDB)
2. Network Routing Service calculation (SSNR)
3. Terminal Proxy Server calculation (SSTR)
4. H.323 Gateway calculation (SSHR)
5. SIP Gateway calculation (SSSR)
6. SIP CTI/TR87
7. PD co-resident with TPS and GW calculation
8. Co-resident Virtual Trunks calculation
9. Signaling Server requirements calculation
10. Alternate and redundant Signaling Server calculation
11. Total number of Signaling Servers calculation

type_index = ISP or CPPM or IBM or HP (Signaling Server type) nrs_type_index = if NRS_Linux = 1 then IBM or HP, else type_index

1. **Signaling Server for Personal Directory/Callers List/Redial List database (SSDB)**

   Personal Directory/Callers List/Redial List (PD/CL/RL) calculations assume that the database resides either on a stand-alone Signaling Server or on a Signaling Server shared with all the other applications. This assumption simplifies the engineering and provisioning rules.

SSDB[type_index]	=	a	if no PD/CL/RL feature
	=	b	if yes on feature, and sharing functions on Signaling Server and IP Phones <= IPL_{DB}[type_index] and HVT+SVT <= VT_{SL}[type_index]
	=	c	if yes on feature, and a stand-alone database Signaling Server or IP Phones > IPL_{DB}[type_index] or HVT+SVT > VT_{SL}[type_index] or dedicated H323 Gateway or SIP Gateway PD dedicated

   if SSDB = c
   SST[type_index] = SST[type_index] +1

2. **Network Routing Service calculation (SSNR)**

   If SIP Proxy required

   Then
   {

   NRS_Linux = 1;

   SSLinux = larger of:

   {

   a NRE ÷ NRE_{l}[nrs_type_index] endpoints (software limit)

   b NRD ÷ NRD_{l}[nrs_type_index] dial plan entries (software limit)

   c NRC ÷ NRC_{HL}[nrs_type_index] calls per hour (hardware limit)

   }

   SSNW_Linux[nrs_type_index] = ROUNDUP(SSLinux) × SPSA(=2, if true; else 1);

   }

   Else {

   NRS_Linux = 0;

   SSNR = larger of:

{

a NRE ÷ NRE_l[nrs_type_index] endpoints (software limit)

b NRD ÷ NRD_l[nrs_type_index] dial plan entries (software limit)

c NRC ÷ NRC_{HL}[nrs_type_index] calls per hour (hardware limit)

d 0 if NRS is not provisioned and there is no branch office

e if (NRE > NRE_{SL}[nrs_index_type] or NRD > NRD_{SL}[nrs_index_type] or NRC > NRC_{SL}[nrs_index_type] then 1

}

NRC can be a hardware, CPU, or memory limit. It includes local NRC_0[nrs_type_index] and network VT_{NET}[nrs_type_index] for this Network Routing Service.
NRC_0[nrs_type_index] is obtained from the main switch calculation.
NRC_{NET}[nrs_type_index] = VT_{NET}[nrs_type_index] × (CCS per VT) × 100 ÷ WAHT ÷ 2
NRC[nrs_type_index] = NRC_0[nrs_type_index] + NRC_{NET}[nrs_type_index]

The calculation for NRC_{NET}[nrs_type_index] requires converting both VT_{323}[nrs_type_index] and VT_{SIP}[nrs_type_index] (from user input) to H.323 and SIP calls. The Signaling Server's capacity for handling SIP calls is different from its capacity for H.323 calls. Therefore, H.323 calls are converted to SIP calls before the load on the Signaling Server is calculated.

Convert H.323 calls to SIP calls by using the ratio of the real-time factors for calls from IP Phones to SIP and H.323 Virtual Trunks: $f_{H/S}$ = (H.323 call real time) ÷ (SIP call real time)

The Signaling Server capacity handling SIP calls is denoted by NRC_{HLS}[nrs_type_index].

Equation (c) in the SSNR calculation becomes:

= [NRC_S[nrs_type_index] + ($f_{H/S}$ × NRC_H[nrs_type_index])] ÷ NRC_{HLS}[nrs_type_index]
where:
NRC_S[nrs_type_index] = the sum of local and network SIP calls the NRS is handling
NRC_H[nrs_type_index] = the sum of local and network H.323 calls the NRS is handling
NRC_{HLS}[nrs_type_index] = the Signaling Server's capacity for handling SIP calls

Equation (c) represents the loading of the Signaling Server for handling NRS calls. It is compared with equations (a) and (b) in order to determine the highest of all potential usages.

If the user wants the Network Routing Service in a dedicated Signaling Server, round up SSNR before proceeding with further calculations:

If SSNR[nrs_type_index] = 0, NRA[nrs_type_index] = No, NRC[nrs_type_index] = 0, NRD[nrs_type_index] = 0, NRE[nrs_type_index] = 0, NRP[nrs_type_index] = 0

if SSNR >= 1 or dedicated NRS required

Then
{

SSNW = ROUNDUP(SSNR) × NRA(=2, if true; else 1)

SST[nrs_type_index] = SST[nrs_type_index] + SSNW

}

Else SS_Co_Res = SS_Co_Res + SSNR

}

3. **Terminal Proxy Server calculation (SSTR)**

SSTR[type_index] = larger of:

{

a	IPL ÷ IPL_{SL}[type_index]	IP Phones software limit
b	C_{IP}[type_index] ÷ IPC_{HL}[type_index]	call limit - calls per hour limit
c	If IPL > IPL_{DB}[type_index] then 1	co-res limit

}

If the user wants Terminal Proxy Server(s) in a dedicated Signaling Server, round up SSTR before proceeding with further calculations:

if SSTR >= 1 or dedicated TPS required

The
n {

SSTW = ROUNDUP(SSTR) + TPSA(=1, if true; else 0)

SST[type_index] = SST[type_index] + SSTW;

}

Else SS_Co_Res = SS_Co_Res + SSTR

}

4. **H.323 Gateway calculation (SSHR)**

SSHR = larger of:

{

a	HVT ÷ HVT_{SL}[type_index]	number of trunks (software limit)
b	Cvt ÷ $HVTC_{HL}$[type_index]	calls per hour (hardware limit)
c	If HVT > VT_{SL}[type_index] then 1	co-res limit

}

If the user wants H.323 Gateway(s) in a dedicated Signaling Server, round up SSHR before proceeding with further calculations:

if SSHR >= 1 or dedicated H323 Gateway is required

The
n {

SSHW = ROUNDUP(SSHR) + GWA(=2, if true; else=1)

SST[type_index] = SST[type_index] + SSHW

}

Else SS_Co_Res = SS_Co_Res + SSHR

}

5. **SIP Gateway calculation (SSSR)**

SSSR = larger of:

{

a	SVT ÷ SVT_{SL}[type_index]	number of trunks (software limit)
b	Cvt ÷ $SVTC_{HL}$[type_index]	calls per hour (hardware limit)
c	SVT > $VTRK_{SL}$[type_index] then 1	co-res limit

}

If the user wants SIP Gateway(s) in a dedicated Signaling Server, round up SSSR[type_index] before proceeding with further calculations:

if SSSR >= 1 or dedicated SIP Gateway is required

The
n {

SSSW = ROUNDUP(SSSR) + GSA(=2, if true; else=1)

SST[type_index] = SST[type_index] + SSSW

}

Else SS_Co_Res = SS_Co_Res + SSSR

}

6. **SIP CTI/TR87 Calculation**

SSTR87 = larger of:

{

a	TR87 ÷ $TR87_{CL}$[type_index]	number of clients (software limit)
b	If HTR87 > $TR87_{SL}$[type_index] then 1	co-res limit

}

If the user wants SIP CTI/TR87 in a dedicated signalling server, then round up SSTR87[type_index] before proceeding with further calculations.

if SSTR87 >= 1 or dedicated SIP CTI is required

The
n {

SSTR87W = ROUNDUP(SSTR87) × TR87A(=2, if true; else=1)

SST[type_index] = SST[type_index] + SSTR87W

}

Else SS_Co_Res = SS_Co_Res + SSTR87

}

7. **PD co-resident with TPS and GW calculation**

If SSDB = b	PD co-res with TPS and GW
{	
If (SSTR+SSHR+SSSR>=1 then	PD cannot co-res with TPS and GW
SST[type_index] = SST[type_index] + 1	add 1 for PD
Co_Res_PD = 0	0 PD not co-res
E lse	
}	

SST[type_index] = SST[type_index] + 1	add 1 for PD and TPS and GW
SS_Co_Res = SS_Co_Res - SSHR - SSSR	remove from co-res
Co_Res_PD = 1	1 PD co-res
If TPSA then SST[type_index] = SST[type_index] + 1	TPS redundant
If GWA or GSA then SST[type_index] = SST[type_index] + 1	H323 or SIP alternate

}

8. Co-resident Virtual Trunks calculation

If SSHR < 1 and SSSR < 1 and Co_Res_PD = 0

Then {

Co_Res_VT = 1	H323 and SIP co-res
If (HVT+SVT > VT_{SL}[type_index]) or (SSHR+SSSR >=1) then	
{	
SST[type_index] = SST[type_index] + 1	Dedicated
SS_Co_Res = SS_Co_Res - SSHR - SSSR	Remove from co-res
If GWA or GSA then SST[type_index] = SST[type_index] + 1	Alternate
Co_Res_VT = 0	VT not co-res
}	
}	

9. Signaling Server requirements calculation

If SS_Co_Res > 1

{

If SSNR > 0 and SSNR < 1 and NRS_Linux = 0

{

SS_Co_Res = SS_Co_Res - SSNR SSNR = 1 SSNW = SSNR × NRA (=2 if true, else 1) SST[type_index] = SST[type_index] + SSNW	If SS_Co_Res > 1 and NRS = co-res application, run NRS on a dedicated Signaling Server
}	
If SS_Co_Res > 1	
{	

Largest = Max of (
If (SSTR > 0 and SSTR < 1 and Co_Res_PD = 0, SSTR, 0),
If (SSHR > 0 and SSHR < 1 and Co_Res_VT = 1 and Co_Res_PD = 0, SSHR, 0),
If (SSSR > 0 and SSSR < 1 and Co_Res_VT = 1 and Co_Res_PD = 0, SSSR, 0),
If (SSTR87 > 0 and SSTR87 < 1, SSTR87 , 0)
	If SS_Co_Res > 1 determine the application with the highest ratio and run that application on a dedicated Signaling Server. Adjust SS_Co_Res with the ratio for that application

)	
If largest = SSTR { SS_Co_Res = SS_Co_Res - SSTR SSTR = 1 SSTW = SSTR × TPSA (=2 if true, else 1) SST[type_index] = SST[type_index] + SSTW }	
If largest = SSHR { SS_Co_Res = SS_Co_Res - SSHR SSHR = 1 SSHW = SSHR × GWA (=2 if true, else 1) SST[type_index] = SST[type_index] + SSHW }	
If largest = SSSR { SS_Co_Res = SS_Co_Res - SSSR SSSR = 1 SSSW = SSSR × GSA (=2 if true, else 1) SST[type_index] = SST[type_index] + SSSW }	

If largest = SSTR87 {

SS_Co_Res = SS_Co_Res - SSTR87

SSTR87 = 1

SSTR87W = SSTR87 × TR87A (=2 if true, else 1)

SST[type_index] = SST[type_index] + SSTR87W }

}

10. **Alternate and redundant Signaling Server calculation**

If SS_Co_Res > 0

Then {

Co_Res_Alt = 0	Determine if alternate is required for co-res
If (SSTR > 0 and SSTR < 1 and Co_Res_PD = 0)	TPS follower
Co_Res_Alt = Co_Res_Alt + 1	
If (SSNR > 0 and SSNR < 1 and NRS_Linux = 0 and NRA = 1) OR	
(SSHR > 0 and SSHR < 1 and Co_Res_VT = ‘ and Co_Res_PD = 0 and GWA = 1) OR	
(SSSR > 0 and SSSR < 1 and Co_Res_VT = 1 and Co_Res_PD = 0 and GSA = 1) OR	
(SSTR87 > 0 and SSTR87 < 1 and TR87A = 1)	
then	
Co_Res_Alt = Co_Res_Alt +1	
}	

11. **Total number of Signaling Servers calculation**

SST[type_index] = SST[type_index] + ROUNDUP(SS_Co_Res) + Co_Res_Alt	all VxWorks Signaling Servers
If NRS_Linux = 1, then SSNW_Linux[nrs_type_index], else 0	all SIP Proxy Servers

Refer to "Signaling Server calculation" (page 261) for a numerical example illustrating the algorithm.

Maximum number of Failsafe Network Routing Services

This algorithm defines the maximum number of Failsafe Network Routing Services (RSF) that can be configured. The maximum RSF is limited by the Primary Network Routing Service (RSP) configuration.

RSF is less than or equal to RSPE
$RSF = (RDE_L \div RSPE) \times [FR - (RFR_S \text{ or } RFR_C)] \times (DDR \div 24) \times (RSP_C)$

Simplified formulas:

RSF = (16 000 ÷ RSPE) for stand-alone Network Routing Service
RSF = (10 000 ÷ RSPE) for collocated Network Routing Service

Table 43 "RSF algorithm constant and variable definitions" (page 250) defines the terms used in the calculations.

Table 43
RSF algorithm constant and variable definitions

Algorithm term	Description	Value	Notes
DDR	Dynamic Data Resynch	24 (Constant that updates with platform changes)	In one day, the minimum number of synchronizations of dynamic data from Active RD to a RSF.
FR	FTP Resource	10 (Constant that updates with system software releases)	Software limit.
RDEL	NRS endpoints limit	5000 (Constant that updates with system software releases)	Software limit.
RSF	Maximum Failsafe NRS allowed	calc (Calculated result)	
RSP_C	RSP CPU performance	1.0 (Constant that updates with platform changes)	PIII 700 MHz; 512 MByte; 20 GByte
RSPE	RSP endpoints	enter (Variable to be entered into the formula)	

Algorithm term	Description	Value	Notes
RFR_C	Reserved FTP Resource Collocated	5 (Constant that updates with system software releases)	Software limit. RSP shares Signaling Server with other applications, such as TPS. Reserve 3 for other applications.
RFR_S	Reserved FTP Resource Standalone	2 (Constant that updates with system software releases)	Software limit. RSP is only application on Signaling Server. Reserve 1 for Static updates and 1 spare.

Reducing imbalances (second round of algorithm calculations)

Input data may not be consistent. For example, there can be a high intraoffice ratio in a call center, or few trunks but a high interoffice ratio. In these cases, the resulting calculations in the EC tool generate a warning message indicating traffic imbalance. The user may want to change the input data and rerun the calculation.

There are two types of imbalances that can occur

- "Virtual Trunks" (page 251)
- "Line and trunk traffic" (page 252)

Virtual Trunks

When the VT number input by the user differs significantly from the calculated VT number (more than 20% difference), the EC tool uses the calculated number and rerun the algorithm to obtain a newer VT number. If the resulting VT number is not stable (in other words, after each rerun, a new calculated VT number is obtained), the program repeats the calculation cycle up to six times. This recalculation looping is built into the EC and occurs automatically, with no user action required. At the end of the recalculation cycle, the user has the choice of using the original input VT number or the final calculated VT number in the configuration.

The user inputs about the number of H.323 Virtual Trunks and SIP Virtual Trunks are a function of other parameters in the system. For example, the number of Virtual Trunks required are affected by the total number of trunks in the system and by intraoffice/incoming ratios: Virtual Trunks and TDM trunks cannot carry a high volume of trunk traffic if the system is characterized as carrying mostly intraoffice calls. If pre-engineering has not provided consistent ratios and CCS, the VT input numbers tend to diverge from the calculated results based on input trunking ratios.

Use the following formula to calculate the VT CCS to compare against user input, in order to determine the size of the deviation. Note that for this consistency check, H.323 VT CCS and SIP VT CCS are separated out from the VT total by using the user input ratio of H.323 to SIP.

$HVT = C_{VT} \times v_H \times WAHT \div 100$
$SVT = C_{VT} \times v_S \times WAHT \div 100$

The respective difference between HVT and HVTCCS, and between SVT and SVT_{CCS}, is the deviation between input data and calculated value.

After the automatic EC recalculations, if the discrepancy between the input VT number and the final calculated number is still significant (more than 20%), follow the recommendations for reducing line and trunk traffic imbalance (see "Line and trunk traffic" (page 252)). Adjusting the number of Virtual Trunks and trunk CCS alone, without changing the intraoffice ratio or its derivatives, may never get to a balanced configuration.

Line and trunk traffic

At the end of the algorithm calculation, if the line and trunk CCS are significantly imbalanced (more than 20% difference), the EC tool generates a warning message. The user can choose whether to change input data and rerun the calculation to have a better balanced system. The recalculation loop starts from the point of entering configuration inputs at the GUI.

Use the following formula to obtain the calculated line CCS to compare against user input, in order to determine the size of the deviation.

Calculated line CCS (LC_{CCS}) = $(C_{SS} + C_{ST} + C_{TS}) \times WAHT \div 100$

The difference between LCCS and LC_{CCS} is the imbalanced line CCS.

Similarly, use the following formula to obtain the calculated trunk CCS to compare against user input, in order to determine the size of the deviation.

Calculated total trunk CCS (TC_{CCS}) = $(C_{TT} + C_{ST} + C_{TS}) \times WAHT \div 100$

The difference between T_{TCCS} and TC_{CCS} is the imbalanced trunk CCS.

Because the calculated CCS factor in traffic ratios, line and trunk CCS can be significantly imbalanced if these ratios are inconsistent. For example, if the intraoffice, incoming, and outgoing ratios are based on contradictory assumptions, the calculated line CCS can be much higher than the number of trunks can absorb.

Table 44 "Tips to reduce traffic imbalances" (page 253) provides tips for users to modify input data so as to steer the algorithm in the right direction. The desired configuration for the input data is when the input numbers for Virtual Trunks, line CCS, and trunk CCS are close to their calculated values (less than 20% difference).

Table 44
Tips to reduce traffic imbalances

When this happens...	Try this...
Line traffic too high	• Reduce CCS per telephone or number of telephones. • Increase the intraoffice ratio.
Trunk traffic too high	• Reduce CCS per trunk or number of trunks. • Reduce the intraoffice ratio. • Increase the tandem ratio (if justified; changing the incoming/outgoing ratio has no impact on line/trunk traffic imbalance).
Need to change input VT number because other input data has changed	• If changing the input VT number is not an option, keep it and change only the number of TDM trunks. • If the input VT number is not a committed value, use the VT number from the previous run. • When input traffic data is changed, expect the resulting VT number to change accordingly. Modify line data or trunk data one at a time to see the trend of convergence. Otherwise, it is hard to know what variable is most responsible for converging to the desired result.

Illustrative engineering example

The following numerical example is for a general business/office model.

Assumptions

The example uses the following values for key parameters.

These parameter values are typical for systems in operation, but the values for the ratios are not the defaults.

- Intraoffice ratio (R_I): 0.35
- Tandem ratio (R_T): 0.03
- Incoming ratio (I): 0.40
- Outgoing ratio (O): 0.22

  In fraction of calls, the above ratios add up to 1.
- AHTSS = 60 [average hold time (AHT) for telephone to telephone ($_{SS}$)]

- AHT_{TS} = 150 [AHT for trunk to telephone ($_{TS}$)]
- AHT_{ST} = 150 [AHT for telephone to trunk ($_{ST}$)]
- AHT_{TT} = 180 [AHT for trunk to trunk ($_{TT}$)]

Given configuration

A CS 1000E system with the following configuration data:

- 1200 digital and analog telephones at 5 CCS/telephone
  - — including 30 TDM ACD agents at 33 CCS/agent
- 1600 IP telephones at 5 CCS/IP telephone
  - — including 60 IP ACD agent telephones at 33 CCS/IP agent telephone
- 410 trunks
  - — including 360 Virtual Trunks (240 H.323 and 120 SIP) at 28 CCS/trunk (The numbers for H.323 and SIP Virtual Trunks are input from user response to a GUI request in the EC.)
- Network Virtual Trunks served by this Gatekeeper: 800 (This is another input from the user interface.)
- CallPilot ports at 26 CCS/CP port
  - — 36 local CallPilot ports
  - — 24 network CallPilot ports (input from user interface)
- Other traffic-insensitive, pre-engineered application ports that require DSP channels and generate real-time feature overhead. For non-blocking configuration, each application port requires one DSP port.
  - — Agent greeting ports: 4
  - — Integrated Conference Bridge ports: 16 (HT = 1800)
  - — Integrated Recorded Announcer ports: 12 (HT = 90)
  - — Integrated Call Assistant ports: 8 (HT = 180)
  - — Hospitality Integrated Voice Service ports: 8 (HT = 90)
  - — Integrated Call Director ports: 12 (HT = 60)
- Features with processing overhead but no hardware ports:
  - — CPND percentage: 20% of TDM telephone calls
  - — Converged Desktop percentage: 5% of the following calls: (intraoffice calls × 0.1) + incoming calls + outgoing calls + tandem calls

— Intraoffice CDR: No (can be yes, but not typical)

— Incoming CDR: Yes

— Outgoing CDR: Yes

— Tandem CDR: Yes

— Symposium-processed ACD calls: 90%

— ACD calls without Symposium: 10%

Real-time factors are based on Table 38 "Real-time factors" (page 223).

Calculations

The calculations in this example were performed by spreadsheet. Some rounding off can have occurred.

- The ACD agent to total telephone ratio = (60 + 30) ÷ (1200 + 1600) = 0.032 This ratio is less than the 15% threshold, so the site is not considered a call center. All ACD traffic is included in call distribution calculations. Refer to "DSP ports for general traffic" (page 230) for more information.
- TDM telephones CCS = [(1200 – 30) × 5] + (30 × 33) = 6840 CCS
- IP telephones CCS = [(1600 – 60) × 5] + (60 × 33) = 9680 CCS
- Fraction of IP calls (P) = 9680 ÷ (6840 + 9680) = 0.586
- Weighted average holding time (WAHT)
  = (60 × 0.35) + (150 × 0.40) + (150 × 0.22) + (180 × 0.03) = 119 seconds
- Total line CCS (L_{CCS}) = 6840 + 9680 = 16 520
- 410 trunks at 28 CCS per trunk:

  Fraction of Virtual Trunks (V) = 360 ÷ 410 = 0.878
  Virtual Trunk traffic (VT_{CCS}) = 360 × 28 = 10 080
  TDM trunk CCS (T_{TDM}) = (410 – 360) × 28 = 1400
  v_H = 240 ÷ (120 + 240) = 0.67
  v_S = 120 ÷ (120 + 240) = 0.33

  Total trunk CCS (T_{TCCS}) = 10 080 + 1400 = 11 480
- Total CCS (T_{CCS}) = 11 480 + 16 520 = 28 000
- Total calls (T_{CALL}) = 0.5 × T_{CCS} × 100 ÷ WAHT
  = 0.5 × 28 000 × 100 ÷ 119 = 11 765
- The system calls are comprised of four different types of traffic: Intraoffice calls (telephone-to-telephone) (C_{SS}); Tandem calls (trunk-to-trunk) (C_{TT}); Originating/Outgoing calls (telephone-to-trunk) (C_{ST}); Terminating/Incoming calls (trunk-to-telephone) (C_{TS}).

1. Intraoffice calls (C_{SS}) = T_{CALL} × R_I

= 11 765 × 0.35 = 4118

a. Intraoffice IP to IP calls (C_{2IP}) = $C_{SS} \times P^2$
= 4118 × 0.586 × 0.586 = 1414
(require no DSP, no VT)
pf1 = 1414 ÷ 11 765 = 0.12

b. Intraoffice IP to TDM calls (C_{1IP}) = $C_{SS} \times 2 \times P \times (1 - P)$
= 4118 × 2 × 0.586 × (1 – 0.586) = 1998
(require DSP)
pf2 = 1998 ÷ 11 765 = 0.17

c. Intraoffice TDM to TDM (C_{NoIP}) = $C_{SS} \times (1 - P)^2$
= 4118 × (1 – 0.586) × (1 – 0.586) = 706
(require 2 DSP, no VT)
pf3 = 706 ÷ 11 765 = 0.06

2. Tandem calls (C_{TT}) = $T_{CALL} \times R_T$
= 11 765 × 0.03 = 353 calls

a. Tandem VT to TDM calls (C_{T1VT}) = $2 \times C_{TT} \times V \times (1 - V)$
= 2 × 353 × 0.878 × (1 – 0.878) = 76
(require DSP and VT)
pf4 = 76 ÷ 11 765 = 0.01

b. Tandem TDM to TDM calls (C_{T2NoVT}) = $C_{TT} \times (1 - V) \times (1 - V)$
= 353 × (1 – 0.878) × (1 – 0.878) = 5
(require 2 DSP, no VT)
pf5 = 5 ÷ 11 765 = 0

c. Tandem VT (H.323) to VT (SIP) calls (C_{T2HS})
= $C_{TT} \times V^2 \times v_H \times v_S \times 2 \times 2$
= 353 × 0.878 × 0.878 × 0.67 × 0.33 × 4 = 242
(require no DSP, VT)
pf6 = 242 ÷ 11 765 = 0.02

3. Originating/outgoing calls (C_{ST}) = $T_{CALL} \times O$
= 11 765 × 0.22 = 2588 calls

a. IP to VT calls (C_{STDI}) = $C_{ST} \times P \times V$
= 235 × 0.586 × 0.878 = 1332
(require VT)
pf7 = 1332 ÷ 11 765 = 0.11

b. IP to TDM calls (C_{STID}) = $C_{ST} \times P \times (1 - V)$
= 2588 × 0.586 × (1 – 0.878) = 185
(require DSP)
pf8 = 185 ÷ 11 765 = 0.02

c. TDM to VT calls (C_{STDV}) = $C_{ST} \times (1 - P) \times (V)$
= 2588 × (1 – 0.586) × 0.878 = 941

(require DSP, VT)
pf9 = 941 ÷ 11 765 = 0.08

d. TDM to TDM calls (C_{STDD}) = CST × (1 – P) × (1 – V)
= 2588 × (1 – 0.586) × (1 – 0.878) = 131
(require 2 DSP, no VT)
pf10 = 131 ÷ 11 765 = 0.01

4. Terminating/incoming calls (C_{TS}) = T_{CALL} × I
= 11 765 × 0.40 = 4706 calls

a. VT to TDM calls (C_{TSVD}) = C_{TS} × V × (1 – P)
= 4706 × 0.878 × (1 – 0.586) = 1711
(require DSP, VT)
pf11 = 1711 ÷ 11 765 = 0.15

b. VT to IP calls (C_{TSVI}) = C_{TS} × V × P
= 4706 × 0.878 × 0.586 = 2421
(require VT)
pf12 = 2421 ÷ 11 765 = 0.21

c. TDM to IP calls (C_{TSDI}) = C_{TS} × (1 – V) × P
= 4706 × (1 – 0.878) × 0.586 = 336
(require DSP)
pf13 = 336 ÷ 11 765 = 0.03

d. TDM to TDM calls (C_{TSDD}) = C_{TS} × (1 – V) × (1 – P)
= 4706 × (1 – 0.878) × (1 – 0.586) = 238
(require 2 DSP, no VT)
pf14 = 238 ÷ 11 765 = 0.02

- From the above data, the real-time multiplier can be obtained:

  Real-time multiplier per call
  = 1 + (f_1 × pf1) + (f_2 × pf2) + (f_3 × pf3) + ... + (f_{14} × pf14) + Error_term
  = 1 + (0 × 0.12) + (2.0 × 0.17) + (0.45 × 0.06) + (2.09 × 0.01) + (2.09 × 0) + (1.93 × 0.02) +(2.27 × 0.11) + (2.42 × 0.02) + (1.93 × 0.08) + (2.27 × 0.01) + (1.46 × 0.15) + (1.6 × 0.21) + (2.00 × 0.03) + (1.2 × 0.02) + 0.25
  = 2.79

- Calls involving at least one IP Phone (needed for Gateway calculation):

  $C_{IP} = C_{2IP} + C_{1IP} + C_{STIV} + C_{STID} + C_{TSVI} + C_{TSDI} = 7686$

- Calls that require DSP resources:

  $C_{DSP} = C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI} = 7405$

- Calls that require Virtual Trunk resources:

  $C_{VT} = C_{T1VT} + C_{T2HS} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} = 6722$

H.323 calls = $C_{VT} \times V_H$ = 6722 × 0.67 = 4504

SIP calls = $C_{VT} \times V_S$ = 6722 × 0.33 = 2218

Real-time calculation with major applications

- ACD agent calls without Symposium = [(Number of ACD agents) × CCS/agent × 100 ÷ AHT_{TS}] × 0.1 × f_{ACD} = 200 × 0.13 = 26
- Symposium calls EBC = [(Number of agents) × CCS/agent × 100 ÷ AHT_{TS}] × 0.9 × fsym = 1994 × 0.9 × 5.7 = 10 229
- Calculate the impact of other major features and applications.

  Application EBC = [(Number of application ports) × CCS per port × 100 ÷ HT] × real-time factor

  Table 45 "Application and feature EBCs and DSP requirements" (page 258) summarizes the EBC calculations. For those applications requiring DSP resources, it also provides the required DSP ports for applications and features, for later use.

  Since DSPs are not a system resource in the CS 1000E, each MG 1000E must provide sufficient DSP resources for non-blocking access to the applications. The rule is to provide one DSP port for each application port in the MG 1000E.

Table 45
Application and feature EBCs and DSP requirements

Application/ Feature	Number of ports	EBC*	Required DSP ports**	Comments
Integrated Conference Bridge	16	37	16	
Integrated Recorded Announcer	12	218	12	
Integrated Call Assistant	8	66	8	
Hospitality Integrated Voice Service	8	132	8	
Integrated Call Director	12	328	12	
Agent greeting	4		4	
CDR - incoming		1506		= 4706 × 0.32
CDR - outgoing		828		= 2588× 0.32
CDR - tandem		155		= 353 × 0.44
*Application EBC = (Number of application ports × CCS per port × 100 ÷ HT) × real-time factor **Required DSP = Number of application ports				

Application/ Feature	Number of ports	EBC*	Required DSP por ts**	Comments
CPND		930		= (1998 + 706 + 1711 + 238) × 0.20, where terminating telephone is a TDM
Converged Desktop		939		= 0.05 × 2.33 × [(4118 × 0.1) + 353 + 2588 + 4706]
Basic ACD		26		
Symposium		10 229		
CallPilot		6474	60	
Total		21 868	120	
*Application EBC = (Number of application ports × CCS per port × 100 ÷ HT) × real-time factor **Required DSP = Number of application ports				

- Add the feature EBC to the system EBC to obtain an accurate estimate of the total CPU load:

  Total system real-time EBC = (Total system calls × real-time multiplier) + Application EBC
  = (11 765 × 2.79) + 21 868 = 54 692

  New system real-time usage

  Compare the total system EBC with the CPU rated capacity to determine the processor utilization.

  CPU utilization = 54 692 ÷ 210 000 = 26.0%

  In this example, CPU utilization, including application and feature impact, is 26.0%. This loading indicates that the CPU can handle this configuration with ease and has plenty of spare capacity.

  DSP calculation for Conference ports

  The formula to calculate the DSP requirement for conference ports is based on the number of telephones in the system. Like application ports, non-blocking conference ports must be provided.

  DSP channels for conference ports = (Number of TDM telephones + Number of IP telephones × 0.028 = 2800 × 0.028 = 79

  DSP calculation without applications

  DSP calls (C_{DSP}) = C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI}
  = 7405

DSP CCS = C_{DSP} × WAHT ÷ 100 = 7405 × 119 ÷ 100 = 8813 CCS

Refer to an Erlang B table (with P.01 GoS) to find the corresponding number of ports, or use the following formula:

Number of DSP ports = DSP CCS ÷ 6192 × 192 = 273

DSP and Media Card calculations

Total DSP ports = DSP for calls + Conference + Applications/features = 273 + 79 + 120 = 472

Number of 32-port Media Cards required = 472 ÷ 32 = 15

For an 8-port Media Card, number of Media Cards required = 472 ÷ 8 = 59

Nortel recommends rounding up the Media Card calculation to an integer.

The numbers calculated for required DSP ports and Media Cards can be considered as a minimum requirement. When the cards are allocated to the Media Gateways, there are placement rules that increase the Media Card requirement (see "Assigning loops and card slots in the CS 1000E" (page 309)).

Virtual Trunk calculation

VT calls (C_{VT}) = C_{T1VT} + C_{T2HS} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} = 6722

H.323 VT calls (HC_{VT}) = C_{VT} × v_H = 6722 × 0.67 = 4504

SIP VT calls (SC_{VT}) = C_{VT} × v_S = 6722 × 0.33 = 2218

VT CCS = C_{VT} × WAHT ÷ 100 = 6722 × 119 ÷ 100 = 8000 CCS

Refer to a Poisson table (with P.01 GoS) to find the corresponding number of trunks, or use the following formula:

Number of Virtual Trunks = VT CCS ÷ 5804 × 192 = 265

Number of H.323 Virtual Trunks = 265 × 0.67 = 178
Number of SIP Virtual Trunks = 265 × 0.33 = 88

User input for number of Virtual Trunks was 360. Since this is greater than 265, it is the number that should be used for further resource calculation.

Signaling Server calculation

The following information was obtained from previous calculations or input data:

Number of IP Phones in the system = 1600
Number of Virtual Trunks = 360 (H.323 = 240; SIP = 120)
Calls involving at least one IP Phone (CIP) = 7686
Calls involving Virtual Trunks (CVT) = GKC0 = 6722

The following is additional user input to the EC tool:

Endpoints served by this NRS: 100
NRS entries (CDP + UDP + É): 1000
Virtual Trunks from other endpoints served by this NRS: 800
NRS alternate (NRA): Yes
TPSA (TPS N+1 redundancy required): Yes
H.323 Gateway alternate (GWA): Yes
SIP Gateway alternate (GSA): Yes
PD/CL/RL feature available to IP Phones: Yes
Sharing database with other traffic: Yes
SIP Proxy: No

1. ***PD/CL/RL database calculation (SSDB)***

   SSDB = b

   The PD/CL/RL feature is available and sharing is allowed.

2. ***Network Routing Service calculation***

   SIP Proxy not required:

   SSNR = larger of:

   {

   a $NRE \div NRE_I = 100 \div 5000 = 0.02$ endpoints

   b $NRD \div NRD_I = 1000 \div 20\,000 = 0.05$ dial plan entries

   c $NRC \div NRC_{HL} = 20706 \div 100000 = 0.21$ calls per hour

   d If ($NRE > NRE_{SL}$ or $NRD > NRD_{SL}$ or $NRC > NRC_{SL}$) then 1 co-res limit
   100 > 100 or 1000 > 1000 or 20706 > 10000 then 1

   }

   NRC_0 is obtained from the main switch calculation.
   $NRC_{NET} = VT_{NET} \times (\text{CCS per VT}) \times 100 \div WAHT \div 2$
   $= 800 \times 28 \times 100 \div 119 \div 2 = 9412$
   $NRC = NRC_0 + NRC_{NET}$

   $f_{H/S}$ = (H.323 call real time) ÷ (SIP call real time) = 1800 ÷ 1200 = 1.5

SIP calls = 120 × 28 × 100 ÷ 119 = 2824
H.323 calls = 240 × 28 × 100 ÷ 119 = 5647

$NRC \div NRC_{HL}$ = [(5647 × 1.5) + 2824 + 9412] ÷ 100 000 = 0.21

This represents the loading of the Signaling Server for handling NRS calls. Compared with the results of equations a, b, c and d: 1 is the highest of all potential usages.

Since SSNR >= 1
SSNW = ROUNDUP(1) × 2 = 2
SST = SST + SSNW = 2

3. ***Terminal Proxy Server calculation***

SSTR = larger of:

{

a $IPL \div IPL_{SL}$ = 1600 ÷ 5000 = 0.32

b $IPC \div IPC_{HL}$ = 7686 ÷ 15 000 = 0.51

c If IPL > 1000 then 1 => 1600 > 1000, 1

}

The larger of the three values is 1.

Since SSHR >= 1:
SSTW = ROUNDUP(1) + 1 = 2
SST = SST + SSTW = 2 + 2 = 4

4. ***H.323 Gateway calculation***

SSHR = larger of:

{

a $HVT \div HVT_{SL}$ = 240 ÷ 1200 = 0.2

b $C_{VT} \div HVTC_{HL}$ = 4504 ÷ 18 000 = 0.25

c HVT > 400 => 240 > 400, 0

}

The larger of the two values is 0.25.
Since SSHR < 1 (0.25 < 1)
SS_Co_Res = SS_Co_Res + SSHR = 0 + 0.25 = 0.25

5. **SIP Gateway calculation (SSSR)**

SSSR = larger of:

{

a $SVT \div SVT_{SL} = 120 \div 1800 = 0.07$

b $C_{VT} \div SVTC_{HL} = 2218 \div 27\,000 = 0.08$

c SVT > 400 => 120 > 400, 0

}

The larger of the two values is 0.08.
Since SSSR < 1 (0.08 < 1):
SS_Co_Res = SS_Co_Res + SSSR = 0.25 + 0.08 = 0.33

6. **SIP CTI/TR87 calculation (SSTR87)**

   No SIP CTI/TR87 specified.

7. **PD co-resident with TPS and GW**

   SSDB = b

   {

   If (SSTR+SSHR+SSSR) >= 1 =>1 + 0.25 + 0.08 + 1.33

   Then

   SST = SST + 1 = 4 + 1 = 5

   Co_Res_PD = 0

   }

8. **Co-resident Virtual Trunk calculation**

   If SSHR < 1 and SSSR < 1 and Co_Res_PD = 0

   0.25 < 1 and 0.08 < 1 and Co_Res_PD = 0 => true

   The
   n

   Co_Res_VT = 1

   If (HVT+SVT VT_{SL} or (SSHR+SSSR >= 1) then

   (240 + 120 > 400) or (0.25 + 0.08 >= 1) => false

9. **Signaling Server requirements calculation**

   If SS_Co_Res > 1 => 0.33 > 1 => false

10. **Alternate and redundant Signaling Server calculation**

    SIP and H323 alternate Gateway required

    Co_Res_Alt = Co_Res_Alt + 1 = 0 + 1 = 1

11. **Total Signaling Server requirement (SST)**

    SST = SST + ROUNDUP(SS_Co_Res) + Co_Res_Alt

= 5 + ROUNDUP(0.33) + 1
= 5 + 1 + 1
= 7

Manual adjustment of Signaling Server requirements

The above calculations strictly followed the engineering rules. However, with a relatively small system such as the one in the example, practical observation and adjustments can allow the user to reduce the number of Signaling Servers required.

1. Having the Personal Directory, Callers List, and Redial List database share functions on a Signaling Server limited the number of IP Phones on that Signaling Server to 1000. With a total of 1600 IP Phones in the system, another Signaling Server was required to provide TPS functions for the additional 600 telephones. In this case, it would be better to put all the IP Phones on the second server, since sharing did not reduce the number of Signaling Servers required.
2. To reduce complexity, the mathematical model allows sharing between the database server and TPS only. Separating the TPS function from the applications database (see item 1) frees up the database server to share with other functions.
3. With 1600 IP Phones, the load on a Signaling Server is 0.32 (1600 ÷ 5000). In this example, the sum of the other three functions (NRS, H.323 Gateway, and SIP Gateway) is 0.53. Combining these three functions with TPS for 1600 IP Phones yields a loading of 0.85. Since this loading is less than 1, it means all four functions can be combined on one Signaling Server.
4. The Signaling Server requirement for this configuration can be restated as follows
    a. 1 Signaling Server for the Personal Directory, Callers List, Redial List applications database
    b. 1 Signaling Server for all other functions (TPS, NRS, H.323/SIP Gateways)
    c. 1 redundant/alternate Signaling Server to cover all four functions.
    d. A total of 3 Signaling Servers for this configuration.

LAN/WAN bandwidth calculation algorithm

The calculation for LAN/WAN bandwidth requirement is based on traffic directly. It does not depend on the traffic model used.

VT traffic in erlangs = [(240 + 120) × 28] ÷ 36 = 280 erlangs

Application engineering

Contents

This section contains information on the following topics:

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"Converged Desktop" (page 266)

"Microsoft Live Communications Server users" (page 275)

"D-channel" (page 281)

"D-channel handler engineering procedure" (page 293)

"CallPilot engineering" (page 299)

"Call Center" (page 300)

"Symposium Call Center" (page 302)

"ELAN engineering" (page 303)

"HSP LAN Engineering" (page 303)

"CLASS network engineering rules" (page 306)

"Configuration parameters" (page 308)

Introduction

Certain applications have significant capacity impact and require engineering in order to operate properly from a capacity perspective. This section provides suggestions for engineering these applications.

For descriptions of the features and their functionality, refer to feature documentation in the Nortel Technical Publications.

For more information on voice networks, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Converged Desktop

The Converged Desktop is a TDM or IP Phone configured to access Multimedia Communication Server (MCS) 5100 multimedia applications through a Session Initiation Protocol (SIP) Virtual Trunk.

Maximum number of Converged Desktop users

In a pure IP system, the CS 1000E can support up to 10 000 Converged Desktop users. However, for a new installation, Nortel recommends configuring no more than 7000 IP users with the Converged Desktop application. This reserves a reasonable amount of real-time capacity for future growth.

SIP access port requirement

Every Converged Desktop call uses a SIP trunk for signaling during the ringing period. In addition, a certain percentage of calls use the SIP trunk for voice traffic for the entire duration of the call. Therefore, the required number of SIP access ports depends on the number of Converged Desktop users and the percentage of voice calls using SIP trunks for conversation.

Personal Call Assistant requirement

The following types of calls to a Converged Desktop use the Personal Call Assistant (PCA) feature for the duration of ringing time:

- calls originating from an internal phone
- calls originating from any non-SIP trunk
- calls originating from a SIP trunk but not from an MCS 5100

The PCA requirement depends only on the number of Converged Desktop users. It is independent of the percentage of voice calls using SIP trunks for conversation.

Calculating SIP access port and PCA requirements

Table 46 "SIP port and PCA requirements for Converged Desktop (with P.01 GoS)" (page 268) shows the required number of SIP access ports and PCAs for different levels of Converged Desktop usage, with P.01 Grade-of-Service (GoS).

The columns under "% voice traffic carried by SIP trunk" indicate the fraction of calls that use a SIP trunk for conversation. A percentage of zero means that the SIP port is used only for signaling during the ringing period and is dropped from the connection once the call is answered.

To use the table, users must know (1) the number of Converged Desktop users and (2) the percentage of Converged Desktop users using SIP trunks to carry voice traffic. The readings below the percentage column are the number of SIP trunks and PCA ports required for a given number of Converged Desktop users.

The number of users shown in Table 46 "SIP port and PCA requirements for Converged Desktop (with P.01 GoS)" (page 268) increments by increasingly large amounts as the number of users increases. If you are calculating requirements for a number of users not shown in the table, use the following formula:

Inputs

- Total number of Converged Desktop users required (MCS_CD_Users)
- Percentage of calls that are answered on a soft client configured as a Converged Desktop (P_CD_SIP)
- Total Number of nonconverged desktop users required (MCS_Non_CD_Users)
- Number of Meet-Me Audio Conference ports configured on the MCS (MeetMe_Ports)

Calculations

- Traffic for CD = (MCS_CD_Users) x (CCS per user) x 10%
- Traffic for SIP ports = (MCS_Non_CD_Users) x (CCS per user) + (MCS_CD_Users x P_CD_SIP) x (CCS per user)
- Total SIP Traffic = (Traffic for CD) x (1 - P_CD_SIP) + (Traffic for SIP ports)
- Number of SIP ports = Poisson (Total SIP Traffic) at P.01 + MeetMe_Ports
- Number of MCS PC As ports = Poisson (Traffic for CD) at P.01
- Number of ACD agents = Number of MCS PCAs ports

If detailed information about the network is not available, use the following formula to estimate the percentage of Converged Desktop users using SIP trunks to carry voice traffic, rounding up the result:

(Number of SIP trunks) ÷ [(Number of SIP trunks) + (Number of H.323 trunks)]

Assumptions

1. The ringing period is 10% of the conversation time. (One ring is a 6-second cycle; the ringing period is usually 2–3 rings; average conversation time is 120–180 seconds.)

2. PCA holding time is equal to the length of the ringing period for each call. This is a conservative assumption, because it implies that every call needs a PCA.

Example

Two hundred Converged Desktop users with 0% SIP trunk conversation require 8 SIP access ports and 8 PCAs. If 20% use SIP for conversation, the requirements are 16 SIP access ports and 8 PCAs.

Table 46
SIP port and PCA requirements for Converged Desktop (with P.01 GoS)

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
25	SIP CCS	12.5	18.1	23.8	29.4	35.0	40.6	46.2	51.9	57.5	63.1	68.8	125.0
	SIP port	3	4	4	4	5	5	5	6	6	6	7	9
	PCA	3	3	3	3	3	3	3	3	3	3	3	3
50	SIP CCS	25.0	36.2	47.5	58.8	70.0	81.2	92.5	103.8	115.0	126.2	137.5	250.0
	SIP port	4	5	6	6	7	7	8	8	9	9	10	15
	PCA	4	4	4	4	4	4	4	4	4	4	4	4
75	SIP CCS	37.5	54.4	71.2	88.1	105.0	121.9	138.8	155.6	172.5	189.4	206.2	375.0
	SIP port	5	6	7	8	8	9	10	11	11	12	13	19
	PCA	5	5	5	5	5	5	5	5	5	5	5	5
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
100	SIP CCS	50.0	72.5	95.0	117.5	140.0	162.5	185.0	207.5	230.0	252.5	275.0	500.0
	SIP port	6	7	8	9	10	11	12	13	14	15	16	24
	PCA	6	6	6	6	6	6	6	6	6	6	6	6
125	SIP CCS	62.5	90.6	118.8	146.9	175.0	203.1	231.2	259.4	287.5	315.6	343.8	625.0
	SIP port	6	8	9	10	12	13	14	15	16	17	18	29
	PCA	6	6	6	6	6	6	6	6	6	6	6	6
150	SIP CCS	75.0	108.8	142.5	176.2	210.0	243.8	277.5	311.2	345.0	378.8	412.5	750.0
	SIP port	7	9	10	12	13	14	16	17	18	20	21	33
	PCA	7	7	7	7	7	7	7	7	7	7	7	7
175	SIP CCS	87.5	126.9	166.2	205.6	245.0	284.4	323.8	363.1	402.5	441.9	481.2	875.0
	SIP port	8	9	11	13	14	16	18	19	20	22	23	37
	PCA	8	8	8	8	8	8	8	8	8	8	8	8
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
200	SIP CCS	100.0	145.0	190.0	235.0	280.0	325.0	370.0	415.0	460.0	505.0	550.0	1000.0
	SIP port	8	10	12	14	16	18	19	21	23	24	26	42
	PCA	8	8	8	8	8	8	8	8	8	8	8	8
225	SIP CCS	112.5	163.1	213.8	264.4	315.0	365.6	416.2	466.9	517.5	568.1	618.8	1125.0
	SIP port	9	11	13	15	17	19	21	23	25	27	28	46
	PCA	9	9	9	9	9	9	9	9	9	9	9	9
250	SIP CCS	125.0	181.2	237.5	293.8	350.0	406.2	462.5	518.8	575.0	631.2	687.5	1250.0
	SIP port	9	12	14	16	19	21	23	25	27	29	31	50
	PCA	9	9	9	9	9	9	9	9	9	9	9	9
300	SIP CCS	150.0	217.5	285.0	352.5	420.0	487.5	555.0	622.5	690.0	757.5	825.0	1500.0
	SIP port	10	13	16	19	21	24	26	28	31	33	36	58
	PCA	10	10	10	10	10	10	10	10	10	10	10	10
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
400	SIP CCS	200.0	290.0	380.0	470.0	560.0	650.0	740.0	830.0	920.0	1010.0	1100.0	2000.0
	SIP port	13	16	20	23	26	29	33	36	39	42	45	74
	PCA	13	13	13	13	13	13	13	13	13	13	13	13
500	SIP CCS	250.0	362.5	475.0	587.5	700.0	812.5	925.0	1037.5	1150.0	1262.5	1375.0	2500.0
	SIP port	15	19	23	27	31	35	39	43	47	50	54	90
	PCA	15	15	15	15	15	15	15	15	15	15	15	15
750	SIP CCS	375.0	543.8	712.5	881.2	1050.0	1218.8	1387.5	1556.2	1725.0	1893.8	2062.5	3750.0
	SIP port	19	26	32	37	43	49	54	60	65	71	76	129
	PCA	19	19	19	19	19	19	19	19	19	19	19	19
1000	SIP CCS	500.0	725.0	950.0	1175.0	1400.0	1625.0	1850.0	2075.0	2300.0	2525.0	2750.0	5000.0
	SIP port	24	32	40	47	55	62	69	77	84	91	98	168
	PCA	24	24	24	24	24	24	24	24	24	24	24	24
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
1250	SIP CCS	625.0	906.2	1187.5	1468.8	1750.0	2031.2	2312.5	2593.8	2875.0	3156.2	3437.5	6250.0
	SIP port	29	38	48	57	66	75	84	93	102	111	120	205
	PCA	29	29	29	29	29	29	29	29	29	29	29	29
1500	SIP CCS	750.0	1087.5	1425.0	1762.5	2100.0	2437.5	2775.0	3112.5	3450.0	3787.5	4125.0	7500.0
	SIP port	33	44	56	67	78	88	99	109	120	130	141	243
	PCA	33	33	33	33	33	33	33	33	33	33	33	33
1750	SIP CCS	875.0	1268.8	1662.5	2056.2	2450.0	2843.8	3237.5	3631.2	4025.0	4418.8	4812.5	8750.0
	SIP port	37	51	63	76	89	101	113	126	138	150	162	280
	PCA	37	37	37	37	37	37	37	37	37	37	37	37
2000	SIP CCS	1000.0	1450.0	1900.0	2350.0	2800.0	3250.0	3700.0	4150.0	4600.0	5050.0	5500.0	10000.0
	SIP port	42	56	71	85	100	114	128	142	155	169	183	318
	PCA	42	42	42	42	42	42	42	42	42	42	42	42
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
2500	SIP CCS	1250.0	1812.5	2375.0	2937.5	3500.0	4062.5	4625.0	5187.5	5750.0	6312.5	6875.0	12500.0
	SIP port	50	68	86	104	121	139	156	173	190	207	224	392
	PCA	50	50	50	50	50	50	50	50	50	50	50	50
3000	SIP CCS	1500.0	2175.0	2850.0	3525.0	4200.0	4875.0	5550.0	6225.0	6900.0	7575.0	8250.0	15000.0
	SIP port	58	80	101	122	143	164	184	205	225	245	266	465
	PCA	58	58	58	58	58	58	58	58	58	58	58	58
3500	SIP CCS	1750.0	2537.5	3325.0	4112.5	4900.0	5687.5	6475.0	7262.5	8050.0	8837.5	9625.0	17500.0
	SIP port	66	91	116	140	165	188	212	236	260	283	307	538
	PCA	66	66	66	66	66	66	66	66	66	66	66	66
4000	SIP CCS	2000.0	2900.0	3800.0	4700.0	5600.0	6500.0	7400.0	8300.0	9200.0	10100.0	11000.0	20000.0
	SIP port	74	103	131	158	186	213	240	267	294	321	347	611
	PCA	74	74	74	74	74	74	74	74	74	74	74	74
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
4500	SIP CCS	2250.0	3262.5	4275.0	5287.5	6300.0	7312.5	8325.0	9337.5	10350	11362.5	12375.0	22500.0
	SIP port	82	114	145	176	207	237	268	298	328	358	388	684
	PCA	82	82	82	82	82	82	82	82	82	82	82	82
5000	SIP CCS	2500	3625	4750	5875	7000	8125	9250	10375	11500	12625	13750	25000
	SIP port	90	125	160	194	228	262	295	329	362	395	428	757
	PCA	90	90	90	90	90	90	90	90	90	90	90	90
6000	SIP CCS	3000	4350	5700	7050	8400	9750	11100	12450	13800	15150	16500	30000
	SIP port	106	148	189	230	270	310	350	390	430	470	509	908
	PCA	106	106	106	106	106	106	106	106	106	106	106	106
7000	SIP CCS	3500	5075	6650	8225	9800	11375	12950	14525	16100	17675	19250	35000
	SIP port	121	170	218	265	312	358	405	451	497	543	589	1057
	PCA	121	121	121	121	121	121	121	121	121	121	121	121
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
8000	SIP CCS	4 000	5 800	7 600	9 400	11 200	13 000	14 800	16 600	18 400	20 200	22 000	40 000
	SIP port	137	192	246	300	353	406	459	512	565	617	669	1205
	PCA	137	137	137	137	137	137	137	137	137	137	137	137
9000	SIP CCS	4 500	6 525	8 550	10 575	12 600	14 625	16 650	18 675	20 700	22 725	24 750	45 000
	SIP port	152	214	274	335	395	454	513	573	632	690	749	1354
	PCA	152	152	152	152	152	152	152	152	152	152	152	152
10 000	SIP CCS	5 000	7 250	9 500	11 750	14 000	16 250	18 500	20 750	23 000	25 250	27 500	50 000
	SIP port	168	236	303	369	436	502	568	633	698	767	834	1502
	PCA	168	168	168	168	168	168	168	168	168	168	168	168
Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.													

Microsoft Live Communications Server users

The Nortel Converged Office feature combines the business-grade telephony of the CS 1000 with the real-time multimedia communication and the remote call control provided by Microsoft® Office Live Communications Server 2005 and Microsoft® Office Communicator 2005 products. Nortel Converged Office is defined by the following two components:

- **Remote Call Control with Session Initiation Protocol (SIP) Computer Telephone Integration (CTI) TR/87** provides full Microsoft® Office integration of telephony to control business-grade telephony phones from within Microsoft® Office applications, as well as support for a standards-based CTI interface defined by the TR/87 protocol.
- **Telephony Gateway and Services** provides a basic SIP Telephony Gateway for connectivity between private and public telephony networks and Live Communications Server 2005 clients.

Trunking

To handle the traffic between the CS 1000 and the Live Communications Server 2005, you must configure sufficient SIP trunks and Personal Call Assistants (PCAs). The number of additional SIP trunks needed is determined by:

The number of Office Communicator users using the SIP Gateway feature.

multiplied by:

The percentage of users expected to be on the phone at any given time.

For example, 100 Office Communicator SIP Gateway users x 10% on the phone at any given time = 10 additional SIP trunks.

The percentage of users on a phone is decided by standard practice and the environment involved (Call Center, Normal Office, and so on).

PCA trunks are required for each Office Communicator user using the twinning (for SIP Gateway) feature.

Calculating SIP access port and PCA requirements

the following table defines the inputs used to calculate SIP access ports and PCA requirements.

Table 47
Inputs

Input	Description
TN_MO_Users	Total number of Office Communicator users that utilize the SIP Access Ports for voice services.
PCA_MO_Users	Number of Office Communicator users that utilize Personal Call Assistant (PCA). The value entered is in addition to the number you indicate on the software screen.
P_PCA_SIP	Percentage of PCA calls that will utilize the soft client to answer.

Calculations

The following formulas are used to calculate traffic requirements:

Traffic for PCAs = (PCA_MO_Users) x (CCS per user) x (1 - P_PCA_SIP) x 10%

Traffic for SIP ports = (TN_MO_Users - PCA_MO_Users) x (CCS per user) + (PCA_MO_Users x P_PCA_SIP) x (CCS per user)

Total SIP Traffic = (Traffic for PCAs) + (Traffic for SIP ports)

Number of MO SIP ports = Poisson (Total SIP Traffic) at P.01 Grade of Service

MO = Microsoft® Office Communicator

Table 48 "Traffic figures" (page 277) shows traffic in CCS and number of ports calculated based on Poisson formula at P.01 Grade of Service.

Table 48
Traffic figures

Traffic (CCS)	Traffic (Erlang)	#Ports
5	0.14	2
10	0.28	3
15	0.42	3
20	0.56	4
25	0.69	4
30	0.83	4
35	0.97	5
40	1.11	5
45	1.25	5
50	1.39	6
55	1.53	6
60	1.67	6
65	1.81	6
70	1.94	7
75	2.08	7
80	2.22	7
85	2.36	7
90	2.5	8
95	2.64	8
100	2.78	8
125	3.47	10
150	4.14	12
175	4.86	13

200	5.56	14
225	6.25	15
250	6.94	16
275	7.64	17
300	8.33	18
325	9.03	19
350	9.72	19
375	10.42	20
400	11.11	21
425	11.81	22
450	12.5	23
475	13.19	24
500	13.89	25
550	15.28	26
600	16.67	28
650	18.06	29
700	19.44	31
750	20.83	33
800	22.22	35
850	23.61	36
900	25	38
950	26.39	40
1000	27.78	42
1500	41.67	58
2000	55.56	74
2500	69.44	90
3000	83.33	106
3500	97.22	121
4000	111.11	137
4500	125	152
5000	138.89	168
6000	166.67	198
7000	194.44	228
8000	222.22	258
9000	250	288

10000	277.78	318
20000	555.56	611
30000	833.33	908
40000	1111.11	1205
50000	1388.89	1502
60000	1666.67	1799
70000	1944.44	2096

SIP CTI/TR87

When planning for capacity with SIP CTI services, observe the following fundamental restriction:

For a single call server that supports multiple nodes, each with SIP CTI services enabled, multiple SIP CTI/TR87 sessions can be established for a given DN through the same node, but not through different nodes.

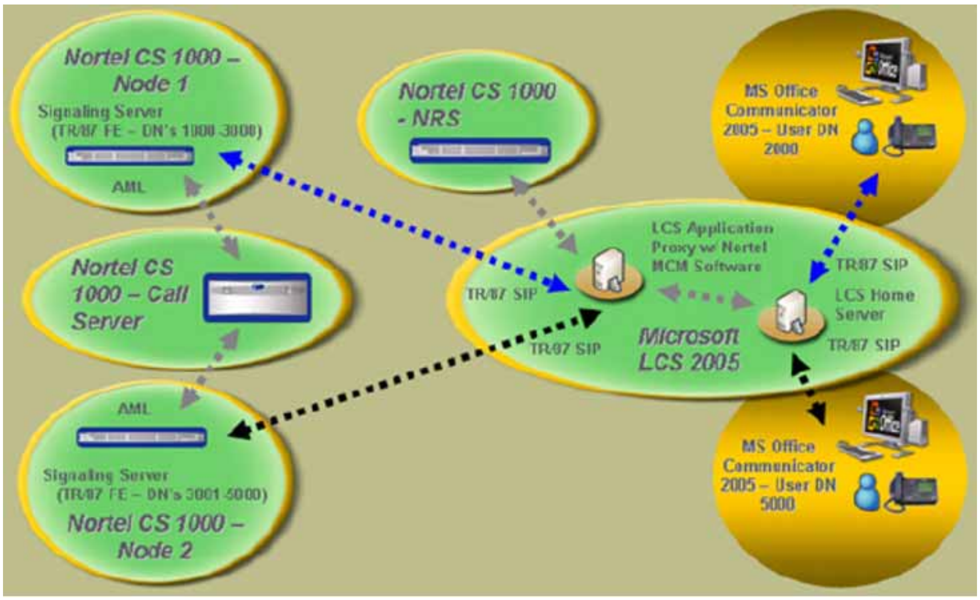
To illustrated this restriction, consider the following high-level example:

Client A sends a TR/87 SIP INVITE to Node 1 to monitor DN 1000. The TR/87 association is established. Client B then sends a TR/87 SIP INVITE to Node 1 (the same node) to monitor DN 1000. Both sessions are established successfully. As a result of this sequence, two TR/87 sessions exist for DN 1000 through Node 1.

However, if Client B attempts to send a TR/87 SIP INVITE to Node 2 (that has an AML link to the same call server as Node 1), the attempt to establish the TR87 sessions fails because the DN is already in use by client A's session through Node 1.

To solve this issue when planning for capacity, SIP routing must ensure that all TR/87 session for a given DN always terminate on the same node when a single Call Server has multiple nodes. (See Figure 57 "Capacity example" (page 280).

Figure 57
Capacity example



This situation can arise in cases where there is an expectation that a single user has multiple clients logged on simultaneously, such as a client at home, a client in the office, and a mobile client all with TR/87 capability.

Impact on Signaling Server The maximum number of SIP CTI/TR87 users on a single Signaling Server is 5000. Signaling Server memory of 1GB is required in the following scenarios:

- SIP CTI/TR87 is co-resident with PD/RL/CL application.
- SIP CTI/TR87 is co-resident with H.323/SIP Gateway serving more than 200 ports, or co-resident with Terminal Proxy Server serving more than 1000 IP users.

Impact on Call Server For different CPUs, the number of supported users is:

- CP PII: 7000 users
- CP PIV: 15000 users
- CP PM: 15000 users

MCM capacity The Standard Performance Evaluation Corporation (SPEC) is a nonprofit corporation formed to establish, maintain, and endorse a standardized set of relevant benchmarks that can be applied to the newest generation of high-performance computers.

Multimedia Convergence Manager (MCM) is a software component designed specifically for the Nortel Converged Office feature to ensure the proper interoperability between Microsoft® and Nortel systems with respect to protocols, users, and phone numbers managed within the Microsoft® Active Directory®.

MCM capacity numbers depend on the hardware platform this application runs on, and the unit used to identify the platform is SPECint.

A single MCM can support 15000 calls per hour (this is a projected value of 3000 users averaging 5 calls per hour; customers should check this with Windows Performance Monitor) for each box, with a SPECint of 13.8.

Because MCM co-resides with Microsoft® Live Communications Server on different platforms, the formula for different hardware platforms is:

Number of calls per hour supported = (15000 x SPECint for a box) / 13.8

The SPECint for each box can be found at www.spec.org

D-channel

D-channel handling interfaces are based on the Multi-purpose Serial Data Link (MSDL) used in Large Systems.

CS 1000E usage of D-channels for digital trunking is the same as the CS 1000M, therefore this section applies to the engineering of D-channels for digital trunking on the CS 1000E.

Engineering considerations

The engineering guidelines assume normal traffic consisting of valid call processing and administrative messages. Engineering rules cannot prevent a piece of equipment on the network from malfunctioning and generating spurious messages, which overload the links. At this point the recovery mechanism becomes essential. The mechanism is graceful, not requiring manual intervention, and can provide as much diagnostic information as possible, to help isolate the root cause of the problem.

Outgoing messages originate from the system Core Processor (CP), are passed to the D-channel handler, and travel across the appropriate link to the destination. In equilibrium, or over a relatively long period of time (on the order of several minutes), the system cannot generate messages faster than the D-channel handler can process them, than the link can transmit them, or than the destination can process them. Otherwise, messages build up at the bottleneck and are eventually lost. The entity with the lowest capacity is the system bottleneck. For very short periods of time, however, one or more entities can be able to send messages at a higher rate than the system bottleneck, since buffers are available to queue the excess messages. These periods are referred to as bursts. The length of the burst and the size of the burst that can be supported depend on the sizes of the buffers.

Thus, to properly engineer a system, two areas must be considered:

- Equilibrium or steady-state performance, which requires an analysis of the CP processing capacity of the various components of the system, along with link bandwidth. The equilibrium analysis assumes 30% peakedness, which is consistent with models for the system CP.
- Burst performance, which requires an analysis of the buffer utilization of the system.

D-channel handling architecture

The D-channel handler and system exchange messages using an SRAM and interrupt scheme. To prevent any one application from tying up buffer resources, a flow control mechanism is defined at the system and D-channel handling interface level. The flow control mechanism is based on the common window mechanism, in which the number of messages outstanding in the transmit or receive direction per socket, or port, cannot exceed T(K) or R(K), respectively. In the transmit direction, for example, a message is considered outstanding from the time the SL-1 software writes it into the transmit ring until all processing of the message by the D-channel handler is completed. Currently T(K) and R(K) are both set at 30. Each application must queue messages if the flow control threshold is exceeded. Typically, the system task also has a buffer for messages.

An overload control threshold is also implemented in the incoming direction to protect the system Core Processor (CP) from excess messages. If the incoming messages on a single port exceed 200 messages in 2 seconds, the port is locked out, and a port overload message is printed. Manual intervention is required to clear the overloaded port. This feature prevents a single port from locking up the whole link.

Several software tasks exist on the D-channel handler. Layer 1 message processing operates at the highest priority. If the link is noisy, Layer 1 processing can starve the Layer 2 and Layer 3 processing tasks, resulting in buffer overflows. If such a problem is suspected, the Protocol Log (PLOG) can be examined. PLOG reporting is requested in LD 96, as described in *Software Input Output Administration (NN43001-611)*.

D-channel

For interfaces including NI-2, Q-SIG, and Euro-ISDN, Layer 3 processing is also performed on the D-channel handler, thus reducing its capacity. These interfaces are referred to as R20+ interfaces. The steady state message rate allowable for D-channel messages is 29 msg/sec for R20+ interfaces.

The SL-1 software output queue for DCH messages is the Output Buffer (OTBF), which is user configurable for between 1 and 127 buffers in LD 17. This is a single system resource shared by all D-channels.

It is possible to define overload thresholds per D-channel for R20+ interfaces. The ISDN_MCNT (ISDN message count), defined in LD 17, specifies the number of ISDN Layer 3 call control messages allowed per 5-second interval. Overload control thresholds can be set per D-channel, ranging from 60 to 350 messages in a 5-second window, with a default of 300 messages. If the overload control threshold is exceeded, DCH421 is output. When the message rate exceeds the threshold for two consecutive 5-second periods, overload control is invoked and new incoming call requests are rejected by the Layer 3 protocol control in the third 5-second time interval. Layer 3 resumes accepting new calls at the end of the third time interval. This flexibility lets the user to regulate the processing required by a specific R20+ DCH port.

The default value implies no overload control, since 300 messages/5 seconds exceeds the rated capacity of 29 messages/second.

Primary Rate Interface network

Equilibrium analysis A D-channel can be configured to support up to 383 B-channels (or 382 with a backup D-channel) on a T1 or 480 B-channels on an E1. The bandwidth available for messages is 64 kbps. Assumptions for a typical application are: 8 messages/call, 29 bytes/message, including 18 bytes of Layer 3 data and 11 bytes of Layer 2 overhead, 28 hundred call seconds (CCS)/trunk, and 180 second Average Hold Time (AHT)/call. The system capacity is derived from its call-carrying capacity for 100% incoming Primary Rate Interface (PRI) calls.

Under the traffic assumptions described above, the D-channel handler is able to support basic call processing messages for 4 D-channels under normal (steady-state) operation.

Peak analysis When there is a link restart, STATUS messages are sent to all trunks with established calls. Since the SL-1 software task does not implement flow control on this mechanism, a burst of up to several hundred messages can be sent to the D-channel handler, exceeding flow control thresholds. When this happens, messages back up on the OTBF buffer, possibly resulting in buffer overflow, as indicated by DCH1030 messages. OTBF overflow is also possible after an initialization, since a burst of messages is sent to each D-channel in the system, and the OTBF is a shared system resource.

The system capacity is significantly higher in this scenario than in the steady state one because it is sending out D-channel messages that do not involve call processing. D-channel handling and Link capacities are also higher because, for equilibrium analysis, some capacity is reserved for peaking.

In the worst case scenario for a single D-channel, if the system sends messages at its peak rate, OTBF buffer overflow is possible. Also, once the messages are sent, a burst of responses can be expected in the incoming direction, resulting in additional congestion at the D-channel handler.

This situation also occurs when a backup D-channel becomes active, since STATUS messages are exchanged to resynchronize the link.

To reduce the possibility of this problem occurring, limit the number of B-channels supported by a D-channel, separate D-channels onto several cards so that message bursts are not being sent to ports on the same D-channel handling card after initialization, and increase the size of OTBF to the maximum value of 127.

The Status Enquiry Message Throttle is implemented. This feature applies only to system-to-system interface networks. It lets the user to configure the number of Status Enquiry messages sent within 128 msec on a per-D-channel basis. The SEMT parameter is set in LD 17 with a range between 1 and 5. The default value is 1. Since this feature provides a flow control mechanism for Status Enquiry messages, the likelihood of buffer overload is reduced.

B-channel overload In an Automatic Call Distribution (ACD) environment, in which the number of ACD agents plus the maximum ACD queue length is considerably less than the number of B-channels available for incoming calls, a burst of incoming messages can impact the performance of the D-channel handler as well as the system via the following mechanism: Calls from the CO terminate on a specified ACD queue. When the destination is busy (the destination telephone is busy or the ACD queue has reached its maximum limit of calls), the system immediately releases the call. The CO immediately presents another call to the same destination, which is released immediately by the PBX, and so on.

The B-channel Overload Control feature addresses this problem by delaying the release of an ISDN PRI call by a user-configurable time when the call encounters a busy condition. The delay in releasing the seized B-channel prevents a new call from being presented on the same B-channel, decreasing the incoming call rate. The timer BCOT is set in LD 16 with a range between 0 and 4000 msec.

ISDN Signaling Link network

In an ISDN Signaling Link (ISL) application, a modem is used to transmit ISDN signaling messages. Baud rates are user configurable at the standard RS232/RS422 rates: 300, 1200, 2400, 4800, 9600, and 19 200 bps (see Table 49 "ISL link capacities" (page 285)). In this case, the modem baud rate constraint can be the limiting constraint. The messages/second that can be supported by the baud rates are given below, where the values allow for 30% peakedness.

The B-channels that can be supported assume the messaging required for a typical application as described in "Equilibrium analysis" (page 283).

Table 49
ISL link capacities

Modem baud rate	Link capacity (msgs/sec)	B-channels that can be supported
300	1 input 1 output	46
1200	4 input 4 output	180
2400	7 input 7 output	316
4800	15 input 15 output	382(T1)/480(E1)
9600	29 input 29 output	382(T1)/480(E1)
19 200	58 input 58 output	382(T1)/480(E1)

For the baud rates listed in Table 49 "ISL link capacities" (page 285), the link is the limiting constraint. The potential peak traffic problems described in "Peak analysis" (page 283) apply here as well, to an even greater extent because of the larger rate mismatch between the system and the system bottleneck. To minimize the risk, set the baud rate as high as possible.

Virtual Network Services network

Concepts applicable to ISL networks also apply to Virtual Network Services (VNS) networks. Up to 4000 VNS DNs (VDN) are supported.

D-channel bit rate The following guidelines provide the basis for engineering the Network ACD (NACD)/VNS D-channel speed.

The bit rate load on the D-channel equals:

> the amount of messages × the octets per message × the number of messages per second

For example, if Facility Message burst is opened with 25 calls in the queue, then the Call Request queue size is greater than or equal to 25. The outgoing facility call request is 25 messages in one second. The incoming facility call request acknowledges 25 messages in the same second. The outgoing and incoming call requests total 50 messages.

In this example, the bit rate load on the D-channel equals:

50 messages × 70 octets × 8 bits/octet
= 28 800 bits/second

Total bandwidth of a 9600 baud modem is approximately:

9600 baud × 2
= 19 200 bits/second

With a total bandwidth of 19 200 bits/second and a bit rate load of 28 800 bits/second, the D-channel cannot handle the messaging. D-channel messaging is backlogged.

If the customer is having problems networking calls during high traffic, then the D-channel can be the cause (especially if the bandwidth is less than 2800 baud). If the D-channel messaging is delayed to the point where VNS call processing gets delayed, the calls fail to network and many PRI/VNS/DCH messages are output at both the source and target nodes.

NACD network

A Network ACD (NACD) network is difficult to engineer, since performance depends on specific network configuration details including connectivity, routing tables, the number of nodes, the number of queues at each node, and calling patterns.

Diverting calls in NACD is controlled by Routing Tables with timers. Calls diverted by NACD can be answered by the Source ACD DN or any one of up to 20 Target ACD DNs. Each Target can have an individual timer defined, from 0 to 1800 seconds. By using ISDN D-channel messaging to queue Call Requests at remote Target ACD DNs, voice calls are not physically diverted until an idle agent is reserved for that call at the remote Target node.

Nortel recommends that the Routing Table be designed so that Call Requests cascade to the network with the timers staggered. The node that is most likely to have available agents should have the smallest timer value. Otherwise Call Requests flood the network, resulting in inefficient use of network and real-time resources.

An Active Target is available to accept NACD calls, while a Closed Target is closed to incoming calls. When calls in the Call Request queue exceed the Call Request Queue Size (CRQS) threshold, the status changes to Closed.

A Status Exchange message is sent from the Target node to the Source ACD DNs indicating the new status. The Target ACD DN remains Closed to further network call requests until the number of calls in the queue is reduced by the Flow Control Threshold (FCTH).

Equilibrium analysis At the source node, for each call queued to the network but not answered, 4 messages are exchanged. For each call queued to the network and answered, 11 messages are exchanged. Likewise, at the target node, a network call that is queued but not answered requires 4 messages, while a call that is queued and answered requires 11 messages. Messages average 31 bytes.

From a single D-channel perspective, the most difficult network topology is a star network that each agent node is connected to a tandem node. All messages to the other nodes are sent across the D-channel connected to the tandem node.

As an example, consider a site with 2000 calls arriving locally during the busy hour. The timers in the Routing Table are staggered so that 1000 are answered locally without being queued to the network, 500 are answered locally after being queued to an average of two network target queues, and 500 are answered in the network after being queued to an average of four network target queues. Meanwhile, 200 Logical Call Requests arrive from the network, of which 100 calls are answered.

For this same network, assume now that the timers in the Routing Table are not staggered; instead, Logical Call Requests are broadcast to the 4 target nodes in the network as soon as calls arrive at the local node. Also assume that a total of 4000 calls arrive elsewhere in the network and are queued at local ACD DNs. Even if the calls are answered exactly where they were before, the number of messages exchanged increases significantly:

- 1500 calls queued on 4 ACD DNs and not answered × 4 msgs/call/DN = 24 000 msgs
- 500 calls answered × 11 msgs/call = 5500 msgs
- 500 calls queued on 3 ACD DNs and not answered × 4 msgs/call/DN = 6000 msgs
- 3900 network calls queued on local DN and not answered × 4 msgs/call = 15 600 msgs
- 100 network calls answered × 11 msgs/call = 1100 msgs
- Total 52 200 msgs/hr
- (52 200 msgs/hr) ÷ (3600 secs/hr) = 14.5 msgs/sec

Peak analysis When the CRQS threshold is reached, the target queue broadcasts messages to the source ACD DNs informing them that it no longer accept calls. The size of this outgoing burst of messages depends on the number of source ACD DNs in the network.

Once the FCTH threshold is reached, another Status Exchange message is sent. At that point, Logical Call Request messages are sent by the Source ACD DNs. While the target queue has been closed, many calls can have queued at source ACD DNs, resulting in a burst of Logical Call Request messages once the DN becomes available.

If CRQS values are set high, many messages are exchanged, with the network emulating a single virtual queue. If the CRQS values are lowered, fewer Call Requests are sent across the network. However, average source delays can be increased. If FCTH levels are set too low, target nodes can bounce between Active and Closed states, resulting in network congestion and excessive real-time utilization. However, if FCTH levels are set too high, a target node can be inundated with Logical Call Request messages once it becomes available. CRQS is configurable for the range 0 to 255, while FCTH is configurable for the range 10 to 100.

Since the impact of these parameters depends on the configuration, it is not possible to make general recommendations on how to configure them. They can be determined as part of the custom network design process. Contact your local Nortel representative for network engineering services.

Impact of proper engineering of B-channels In the NACD environment, another problem arises when insufficient B-channels are configured across the network. When an agent becomes available, an Agent Free Notification message is sent to the source node. An ISDN Call Setup message is sent from the source node to the target node. Since no B-channel is available, the agent reservation timer expires, an ISDN Cancellation Message is sent from the target node to the source node, and an ISDN Cancellation Acknowledge message is sent from the source node to the target node. At this point, the agent is still free, so the process repeats until a trunk becomes available or the target closes. This scenario results in a significant amount of message passing.

Trunk requirements under Longest Idle Agent routing

Trunk requirements are usually calculated using the NACD engineering guidelines, whereby call loading for each queue at each site is estimated and used to calculate the required number of trunks between each pair of sites. However, when Longest Idle Agent (LIA) is used as the routing criterion, load estimation becomes difficult. Assuming that any agent can take any call and that agents have equal holding time characteristics, the following procedure provides a method to estimate the number of trunks required between pairs of sites.

Assumptions

1. All agents reside in one common pool and process calls at an equal rate (in other words, they have a common average call service time).
2. An agent having the longest idle time occurs with equal probability among all of the agents during normal operation.
3. Agents appear as one large pool to incoming calls.

With these assumptions, under LIA, calls are routed proportional to the number of active agents at each site.

Calculation steps

1. Note the number of active agents at each site (ni) and the total number of active agents over all sites (N).
2. Calculate the proportion of active agents at each site:
   pi = ni/N
3. For each incoming local call arrival stream to site i (Ai, expressed in CPH), calculate the calls routed from site i to site j:
   Cij = Ai x pj
4. Calculate the total calls routed (T, expressed in CPH) between each pair of sites:
   Tij = Tji = Cij + Cji
5. Apply Erlang B to each Tij, i < j, to get the number of required trunks between sites i and j (Lij).

   Erlang B requires the following parameters:

   a. Grade-of-Service (GoS) — probability of a blocked call (in other words, no trunk available) — taken to be 0.01

   b. Mean Call Service Time (usually in seconds)

   c. number of calls per hour (CPH)

Refer to "Trunk traffic Erlang B with P.01 Grade-of-Service" (page 336) for values for Erlang B.

Parameter settings

The following are parameters that can be configured in LD 17 for CS 1000 D-channels. Items are listed with their input ranges, with default values shown in brackets.

1. OTBF 1 - (32) - 127: Size of output buffer for DCH

   This parameter configures how many output buffers are allocated for DCH messages outgoing from the system CP to the D-channel handling card. The more that are created, the deeper the buffering. For systems with extensive D-channel messaging, such as call centers using NACD, the parameter can be set at 127. For other systems with moderate levels of D-channel messaging, OTBF can be set at the smaller of the following two quantities: Total B-channels – (30 × MSDL cards with D-channels) or 127.

For example, if a system in a standard office environment is configured with 7 T1 spans, 2 D-channels located on two different NTBK51 daughterboards, and 2 back-up D-channels, the total number of B-channels is (7 × 24) – 4 = 164. OTBF can be configured to be the smaller of 164 – (30 × 2) = 104 and 127 which is 104.

2. T200 2 - (3) - 40: Maximum time for acknowledgment of frame (units of 0.5 secs)

   This timer defines how long the D-channel handler's Layer 2 LAPD waits before it retransmits a frame. It if does not receive an acknowledgment from the far end for a given frame before this timer expires, it retransmits a frame. Setting this value too low can cause unnecessary retransmissions. The default of 1.5 seconds is long enough for most land connections. Special connections, over radio, for instance, can require higher values.

3. T203 2 - (10) - 40: Link Idle Timer (units of seconds)

   This timer defines how long the Layer 2 LAPD waits without receiving any frames from the far end. If no frames are received for a period of T203 seconds, the Layer 2 sends a frame to the other side to check that the far end is still alive. The expiration of this timer causes the periodic "RR" or Receiver Ready to be sent across an idle link. Setting this value too low causes unnecessary traffic on an idle link. However, setting the value too high delays the system from detecting that the far end has dropped the link and initiating the recovery process. The value can be higher than T200. It can also be coordinated with the far end so that one end does not use a small value while the other end uses a large value.

4. N200 1 - (3) - 8: Maximum Number of Retransmissions

   This value defines how many times the Layer 2 resends a frame if it does not receive an acknowledgment from the far end. Every time a frame is sent by Layer 2, it expects to receive an acknowledgment. If it does not receive the acknowledgment, it retransmits the frame N200 times before attempting link recovery action. The default (3) is a standard number of retransmissions and is enough for a good link to accommodate occasional noise on the link. If the link is bad, increasing N200 can keep the D-channel up longer, but in general this is not recommended.

5. N201 4 - (260): Maximum Number of Octets (bytes) in the Information Field

This value defines the maximum I-frame (Info frame) size. There is no reason to reduce the number from the default value unless the system is connected to a system that does not support the 260-byte I-frame.

6. K 1 - (7): Maximum number of outstanding frames

   This value defines the window size used by the Layer 2 state machine. The default value of 7 means that the Layer 2 state machine sends up to 7 frames out to the link before it stops and requires an acknowledgment for at least one of the frames. A larger window allows for more efficient transmission. Ideally, the Layer 2 receives an acknowledgment for a message before reaching the K value so that it can send a constant stream of messages. The disadvantage of a large K value is that more frames must be retransmitted if an acknowledgment is not received. The default value of 7 should be sufficient for all applications. The K value must be the same for both sides of the link.

7. ISDN_MCNT (ISDN Message Count) 60 - (300) - 350: Layer 3 call control messages per 5-second interval

   It is possible to define overload thresholds for interfaces on a per-D-channel basis. This flexibility lets the user to regulate the D-channel handler processing required by a specific R20+ DCH port. The default value of 300 messages/5 seconds is equivalent to allowing a single port to utilize the full real-time capacity of the D-channel handler. To limit the real-time utilization of a single R20+ DCH port to (1 ÷ n) of the real-time capacity of the D-channel handler, for n > 1, set ISDN_MCNT to (300 ÷ n) × 1.2, where the 1.2 factor accounts for the fact that peak periods on different ports are unlikely to occur simultaneously. For example, to limit a single port to one-third of the processing capacity of the D-channel handler, ISDN_MCNT is set to (300 ÷ 3) × 1.2 = 120.

If the ISDN_MCNT threshold is exceeded for one 5-second period, error message DCH421 is printed. If the threshold is exceeded for two consecutive periods, incoming call requests arriving in the third 5-second interval are rejected by the D-channel handler Layer 3 software. At the end of the third 5-second interval, Layer 3 resumes accepting incoming call requests.

Serial Data Interface (SDI)

The SDI ports on the Media Gateway Controller (MGC) and Small System Controller (SSC) cards in the Media Gateways and on the Terminal Server provide an asynchronous serial data interface to TTYs, printers, modems, CRTs, ACD-C package displays and reports, and CDR TTYs.

Normally, in the output direction, the SDI Application passes any character received from the system to the Layer 1 Driver to be sent out over the interface. If XON/XOFF Handling is enabled for printing, the SDI Application buffers up to 500 characters once an XOFF is received. The system is not aware that an XOFF has been received. After the buffer is full, if further output is received, the oldest data is discarded. Output resumes when an XON is received or 1 minute has passed since the output was halted by an XOFF. At this point, the contents in the buffer is emptied first, followed by output from the system. If any data has been discarded, an error message is sent.

In the input direction, every character received by the Layer 1 Driver is passed to the SDI Application. The SDI Application echos any input character unless it is told not to by the system. In Line Editing Mode, the SDI Application buffers a line of up to 80 characters that can be edited before being sent to the system.

Under certain conditions, control characters can cause messages to bounce between a modem or printer and the system. To avoid these situations, configure modems in dumb mode and disable printer flow control.

The system input buffer is the TTY input buffer, which can store 512 characters. The system output buffer is the TTY output buffer, which can store 2048 characters.

Call Detail Recording records

Call Detail Recording (CDR) records are available in two formats: *FCDR=old* and *FCDR=new*. A typical record for the old format is 100 bytes long while a typical record for the new format is 213 bytes long (see Table 50 "Link capacities for CDR application (outgoing)" (page 292)). Due to the nature of the SDI interface, characters are output one at a time, resulting in 100 messages and 213 messages generated for *FCDR=old* and *FCDR=new*, respectively. Each message requires 10 bits. Based on real-time measurements, the MSDL rated capacity for processing CDR messages is 16 631 messages/second.

Table 50
Link capacities for CDR application (outgoing)

Modem baud rate	Link capacity (msg/sec) (peak)	Calls/Hour for *FCDR=old*	Call/Hour for *FCDR=new*
300	30	831	390
1200	120	3323	1560
2400	240	6646	3120
4800	480	13 292	6241
9600	960	26 585	12 481
19 200	1920	53 169	24 962
38 400	3840	106 338	49 924

Equilibrium analysis The system capacity for messages per second is conservatively based on the assumption of 100% outgoing calls with *FCDR=new*. Typically, CDR records are not generated for 100% of the calls.

Peak analysis Since each character is sent as a separate message, every time a CDR record is sent, a traffic peak is generated.

To prevent system buffers from building up, set the baud rate at 38 400. If a lower baud rate is chosen, assume that the CDR application frequently is in a state of flow control. Note that this is true even if the steady state message rate is low, due to the nature of the SDI interface.

The burst sizes are even greater if CDR is configured with queue records for incoming ACD calls.

D-channel handler engineering procedure

It is important to engineer the D-channel handler in the context of engineering the entire system. Refer to *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for additional information on real-time engineering of the system. In all cases with a user configurable link rate, it is essential that the link be configured so that the rate is high enough to support steady-state requirements and some peakedness. Otherwise, the application messages occupy system buffers, increasing the chance of buffer overflow.

Table 51 "D-channel handler engineering worksheet" (page 293) is a high-level worksheet for analysis of D-channel handling capacity. See Table 57 "Real-time requirements for D-channel applications" (page 298) through Table 55 "Peak buffer requirements for SDI applications" (page 296) for the values to use in the worksheet.

Table 51
D-channel handler engineering worksheet

Port	Application	Real Time required	Peak Buffer usage outgoing	Peak Buffer usage incoming
0	____________	________	________	________
1	____________	________	________	________
2	____________	________	________	________
3	____________	________	________	________
Total		________	________	________

Assuming 30% peakedness for the applications, the total real time required should be less than 2 770 000 msec. The projected real-time utilization of the D-channel handler is given by:

Real-time usage = Total Real Time Required ÷ 2 770 000

Nortel recommends that peak buffer usage be less than 60 in each direction. As the peak buffer usage increases over 60, the likelihood of an intermittent buffer-full problem increases.

The following sections provide procedures, including worksheet tables, for calculating the real time required on the D-channel handler for various applications.

In Table 51 "D-channel handler engineering worksheet" (page 293) through Table 55 "Peak buffer requirements for SDI applications" (page 296), if the calls/hour value is known, insert that value into Column A. Otherwise, follow the guidelines provided. Values in parentheses are default values. For example, the default number of calls/hr/trunk is 15.6. The value in Column E can be inserted in the Real Time Required column of Table 51 "D-channel handler engineering worksheet" (page 293), and the appropriate Peak Buffer Usage values should be inserted in the corresponding Peak Buffer Usage columns of Table 51 "D-channel handler engineering worksheet" (page 293).

DCH applications

If several applications share a D-channel, add the final real-time requirements for the applications and then enter the total in the appropriate entry in Table 57 "Real-time requirements for D-channel applications" (page 298).

Table 52
Real-time requirements for D-channel applications

DCH	Calls/hr A	Msgs/call B	Msgs/hr C = A × B	Msec/msg D	Msec E = C × D
ISDN Network	trunks/DCH × calls/hr/trunk (15.6) = ______	8	______	pre-R20: 8.8 R20+: 26.5	______
For clarification of the terms "pre-R20" and "R20+," refer to "D-channel" (page 282)					

DCH	Calls/hr A	Msgs/call B	Msgs/hr C = A × B	Msec/msg D	Msec E = C × D
NACD	NACD agents × calls/hr/agent (18.3) = _____	30	_____	pre-R20: 8.8	_____
NMS	NMS ports × calls/hr/port (65) = _____	10	_____	pre_R20: 8.8	_____
For clarification of the terms "pre-R20" and "R20+," refer to "D-channel" (page 282)					

The calculations described for NACD provide a simplified approximation of a "typical" NACD network. If call flows can be predicted or estimated, they can be used to develop a more accurate model using the number of messages. When this is done, the msgs/hr is computed directly, so columns A and B are not used. See "Examples" (page 297) for a detailed example of how this can be done.

If a live system is being modeled, add the "number of all incoming messages received on the D-channel" and the "number of all outgoing messages sent on the D-channel" field from a busy hour TFS009 report to derive the entry for Column C. See *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for details.

Table 53
Peak buffer requirements for D-channel applications

DCH	Outgoing	Incoming
ISDN Network	SEMT (1) × 8	SEMT (1) × 8
NACD	Source ACD DNs + 5 = ____	Network congestion level: • Low: 10 • Medium: 20 • High: 30
NMS	10	10

In the case of an ISL D-channel, ensure that the baud rate of the connection is greater than

(C msgs/hr × 29 bytes/msg × 8 bits/byte) ÷ 3600 sec/hr

where C comes from column C in Table 57 "Real-time requirements for D-channel applications" (page 298).

If the baud rate is too low to meet requirements, performance of the entire D-channel handler can be jeopardized, since 30 of the output buffers are occupied with ISL D-channel messages and the real time spent processing these messages increases due to additional flow control and queueing logic.

SDI applications

In the HSL analysis, include live agents, automated agents, and CallPilot agents in the agent total. This compensates for the assumption of simple calls.

Table 54
Real-time requirements for SDI applications

SDI	calls/hr A	msgs/call B	msgs/hr C=A×B	msec/msg D	msec E=C ×D
CDR	calls/hr with reports = ______	FCDR = old:100 FCDR = new: 213	______	0.05	______
HSL	agents × calls/agent/hr (18.3) = ______	5	______	8.8	______
TTY	NA	NA	15 000	0.05	______

There are no traffic reports that provide information on the number of SDI messages directly. For CDR records, determine whether CDR is enabled for incoming, outgoing, and/or internal calls. The number of incoming, outgoing, internal, and tandem calls is available from TFC001. Tandem calls are considered both incoming and outgoing. Alternatively, the number of CDR records can be counted directly.

Table 55
Peak buffer requirements for SDI applications

SDI	Outgoing	Incoming	Minimum baud rate
CDR	• 30 if baud rate is less than recommended in Table 50 "Link capacities for CDR application (outgoing)" (page 292)	1	(msgs/hr × 10 bits/msg) ÷ (3600 sec/hr) = _____
HSL	• Messages per call — simple: 5 — medium: 10 — complex: 15	1	(msgs/hr × 20 bytes/msg × 9 bits/byte) ÷ (3600 sec/hr) = _____
TTY	10	10	

Examples

NACD network with CDR reports

Consider an NACD network with the topology given in Figure 58 "NACD network" (page 297). The call flow is provided, where arrows indicate where calls enter the network and where they are answered.

Each node has a single ACD DN and calls are queued to the network target DNs as soon as they arrive.

For this network, we wish to determine whether a single D-channel handler on Node B can support DCH1, DCH2, and an SDI port for CDR records on Port 0.

Since we have detailed call flow information, we can develop a messaging model for DCH1 and DCH2 (see Table 56 "NACD Message Model" (page 297)).

Figure 58
NACD network

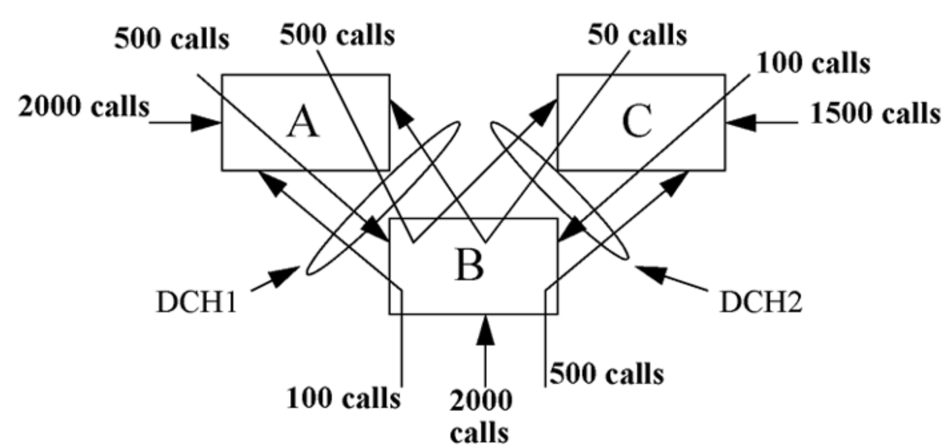


Table 56
NACD Message Model

Originating Node	Total Queued	Queued and answered	Queued but not answered	Total messages	DCH1	DCH2
Node A to Node B	3000	500	2500	15 500	x	x
Node A to Node C	3000	500	2500	15 500	x	x
Node B to Node A	2600	100	2500	11 100	x	

Originating Node	Total Queued	Queued and answered	Queued but not answered	Total messages	DCH1	DCH2
Node B to Node C	2600	500	2100	13 900		x
Node C to Node A	1650	50	1600	6950	x	x
Node C to Node B	1650	100	1550	7300	x	x

The DCH1 and DCH2 columns indicate whether the messages can be included in the DCH1 and DCH2 message count, respectively. For each row, multiply the entry in the "Queued and answered" column by 11 messages and multiply the entry in the "Queued but not answered" column by 4 messages. The sum of these two values is provided in the "Total messages" column. By summing the rows that can be included for DCH1 and DCH2, we derive the total messages for DCH1: 56 350 msg/hr and DCH2: 59 150 msg/hr. Note that these messages do not include the impact of CRQS and FCTH, which are beyond the scope of this analysis (see Table 57 "Real-time requirements for D-channel applications" (page 298)).

Table 57
Real-time requirements for D-channel applications

DCH	calls/hr A	msgs/call B	msgs/hr C=A×B	msec/msg D	msec E=C×D
NACD DCH1	NA	NA	56 350	pre-R20: 8.8	495 880
NACD DCH2	NA	NA	59 150	pre-R20: 8.8	520 520

Assuming that no non-NACD calls are carried, Node B carries 3750 calls/hour.

Table 58
Real-time requirements for SDI applications

SDI	calls/hr A	msgs/call B	msgs/hr C=A×B	msec/ msg D	msec E=C ×D
CDR	calls/hr with reports=3 750	FCDR=old: 100 FCDR=new: 213	798 750 (FCDR=new)	0.05	39 938

The total D-channel handler requirements can then be computed:

Table 59
Engineering worksheet

Port	Application	Real Time required	Peak Buffer usage outgoing	Peak Buffer usage incoming
0	CDR	39 938	10	1
1	DCH-NACD	495 880	7	10
2	DCH-NACD	520 520	7	10
3				
Total		1 056 338	24	21

The projected D-channel handler utilization is 1 056 338 ÷ 2 770 000 = 38%. Assuming low network congestion, incoming and outgoing peak buffer usage are below 60, so a single D-channel handler is able to support this configuration. However, due to the potentially high messaging impact of NACD, this can be re-engineered periodically to determine whether the call volumes or call flow patterns have changed.

CallPilot engineering

Refer to *CallPilot Planning and Engineering (555-7101-101)* for engineering details. The abbreviated procedure in this chapter is for system engineering where a rough estimate of CallPilot ports (or channels) is required.

In addition to voice channels, a CallPilot allows fax and speech-recognition media. As a measure of Digital Signal Processing (DSP) power, different media types require different Multimedia Processing Unit (MPU) quantities:

- One voice channel requires one MPU.
- One fax channel requires two MPUs.
- One speech-recognition channel requires four MPUs.

A Multimedia Processing Card (MPC-8) is a credit-card sized PC card that resides in the CallPilot Server. Each MPC-8 has eight MPUs. The maximum number of MPUs in a CallPilot is 96. Any use of non-voice application reduces the number of channels available for voice traffic.

For an IP source to access CallPilot, the codec must be set for G.711. Since a non-standard proprietary codec is used in CallPilot, a multi-rate transcoding renders the resulting voice samples with very poor quality.

The default holding time for a voice channel user is 40 seconds in the CallPilot port engineering. Another resource to be estimated in CallPilot is storage size. This requires a complicated calculation and is not be covered here. Refer to *CallPilot Planning and Engineering (555-7101-101)* for details.

Once the CCS for each type of media is calculated, sum up the total and refer to capacity tables in the NTP for the MPU requirement based on the offered CCS traffic.

For non-blocking access, provide one DSP port for each CallPilot port equipped.

Call Center

The Call Center is an ACD switch whose calls are mostly incoming, with extensive applications features such as Nortel Hospitality Integrated Voice Services. A port in the Call Center environment, either as an agent telephone or trunk, tends to be more heavily loaded than other types of applications.

System capacity requirements depend on customer application requirements, such as calls processed in a busy hour, and feature suites such as Recorded Announcement (RAN), Music, and Interactive Voice Response (IVR).

ACD

Automatic Call Distribution (ACD) is an optional feature available with the system. It is used by organizations where the calls received are for a service rather than a specific person.

For basic ACD, incoming calls are handled on a first-come, first-served basis and are distributed among the available agents. The agent that has been idle the longest is presented with the first call. This ensures an equitable distribution of incoming calls among agents.

The system is managed or supervised by supervisors who have access to the ACD information through a video display terminal. These supervisors deal with agent-customer transactions and the distribution of incoming calls among agents.

Many sophisticated control mechanisms have been built on the basic ACD features. Various packages of ACD features have real-time impact on the system CP capacity.

ACD-C1 and C2 packages

ACD Management Reporting provides the ACD customer with timely and accurate statistics relevant to the ACD operation. These statistics form periodic printed reports and ongoing status displays so the customer can monitor changing ACD traffic loads and levels of service and implement corrective action where required.

The ACD-C1 package primarily provides status reporting of the system through a TTY terminal. To control and alter the configuration of the system, the ACD-C2 package is required; it provides the load management commands. The following is a partial list of functions of a supervisor position in the C2 package:

- Assign auto-terminating ACD trunk routes.
- Assign priority status to ACD trunks.
- Reassign ACD agent positions to other ACD DNs.
- telephone the timers and routes for first and second RAN.
- Define the overflow thresholds.
- Specify a night RAN route.

ACD-D package

The ACD-D system is designed to serve customers whose ACD operation requires sophisticated management reporting and load management capabilities. It has an enhanced management display, as the system is supplemented by an auxiliary data system. The system and the auxiliary processor are connected by data links through SDI ports for communications. Call processing and service management functions are split between the system and the auxiliary processor.

ACD-MAX

ACD-MAX offers a customer managerial control over the ACD operation by providing past performance reporting and current performance displays. It is connected through an SDI port to communicate with the system CP. The ACD-MAX feature makes the necessary calculations of data received from the system to produce ACD report data for current and past performance reports. Every 30 seconds, ACD-MAX takes the last 10 minutes of performance data and uses it to generate statistics for the current performance displays. The accumulated past performance report data is stored on disk every 30 minutes.

ACD-MAX calls impact capacity engineering in the real-time area only.

NACD

The majority of tasks in the engineering of Network ACD (NACD) involve the design of an NACD routing table and the engineering of overflow traffic. The process is too complex to be included here. The engineering procedure in this NTP is for single-node capacity engineering, which accounts for the real-time impact of NACD calls on a switch either as a source node or remote target node. Therefore, the overall design of a network is not in the scope of this document.

RAN and Music

The RAN trunk can be treated just like a normal trunk. The only potential capacity impact is for systems that include RAN trunks in blocking or non-blocking calculations. The calculations determine the total number of loops or card slots required.

Music Broadcast requires any Music trunk and an external music source or a Nortel Integrated Recorded Announcer card. The Integrated Recorded Announcer has the capability to provide audio input for external music. A Conference loop is not required for Music Broadcast.

Refer to "Service loops and circuits" (page 196) for more information.

Symposium Call Center

Symposium is a Host Server that interfaces through an Ethernet to enable the system to provide advanced Call Center features to users. Although Internet Protocol (IP) is used for communications, the underlying message to the system input queue is an Application Module Link (AML) message.

The customer can create simple-to-write scripts in Symposium to control processing of an arriving call that is eventually delivered to an agent queue after following various call processing rules, such as skill set of agent, call priority, and length of waiting time.

The complexity of call handling on the system call processor determines the impact of Symposium Call Center on the system. Depending on the script used, the call processing can include giving RAN, Music, and IVR, all of which require a voice-processing system such as CallPilot.

Symposium Call Center with IP phones and Virtual Trunks

When IP Phones are used as ACD agent telephones, there are certain special engineering rules. The following two additional resources must be engineered:

- Digital Signal Processor (DSP) channels (therefore, Media Cards)
- Virtual Trunks

For non-blocking access, provide one DSP port for each ACD agent configured.

Refer to "Resource calculations" (page 215) for the detailed calculations.

ELAN engineering

The Embedded Local Area Network (ELAN) subnet is designed to handle messaging traffic between the system and its applications, such as Symposium and CallPilot. It is not meant to handle functions of the customer's LAN, which carries customer application traffic.

A 64 kbps link can handle messaging traffic of over 80 000 calls. The ELAN subnet, being an Ethernet with data rate of 10/100/1000MG auto-negotiate, is not a bottleneck in a Symposium/CallPilot configuration. However, observe the following engineering guidelines to avoid performance problems. For more detailed information, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

- Ensure that settings on the physical interface of the system to the Ethernet are correct.
- Although no traffic engineering is required on the ELAN subnet, if the loading on the link is extremely high (for example, above 10% on the 10T-10 Mbps), collision on the Ethernet can happen. Use a sniffer to detect any performance problems. Decrease the loading on the link if it is overloaded.
- Set a consistent data rate with the application.

Certain remote maintenance applications can utilize the ELAN subnet to access the system from a remote location. Ensure that no other customer LAN traffic is introduced.

HSP LAN Engineering

The High Speed Pipe (HSP) is used to connect two Call Server CPUs in a Campus Redundant environment. The HSP is used to shadow disk and memory information from one CPU to another and to provide heartbeat information (including health information) from one CPU to the other.

Due to the mission critical role that the HSP provides between the active and redundant Call Server, the HSP must be carefully engineered. This section describes the rules governing the engineering of the HSP. For a more detailed description of how the Campus Redundancy feature works, refer to *System Redundancy Fundamentals (NN43001-507)*.

The HSP can be connected using a cable directly between the two CPUs, or using networking equipment. CP PIV requires the use of a crossover cable for HSP. When using networking equipment to connect, the HSP ports are assigned unique IP addresses.

The following are recommendations and rules for configuring the HSP network and network interfaces of two Call Server CPUs using network equipment:

- The HSP must be connected through an Ethernet cable (cross-over on CP PIV) or by a dedicated VLAN through switches.
- The HSP must be in its own IP subnet. It cannot be combined with the ELAN subnet.
- The minimum throughput of the HSP must be 100 Mbps. Therefore, the HSP port must be 100 Mbps and full duplex. This must be confirmed using the **STAT HSP** command in LD 137 after the equipment is operational. This must also be verified on the network equipment that the HSP is attached.
- The network switches must be capable of port mapping to 802.1p/Q.
- When running the HSP across network equipment, the HSP must be isolated in its own VLAN. Do not include other traffic in this VLAN. This VLAN must be given higher VLAN priority than any other traffic on the network, except for network control traffic (network control traffic is the traffic necessary to keep the network operational). The VLAN must be 802.1p/Q-capable and must be set to a very high setting so as not to starve the HSP. Nortel strongly recommends 802.1p Level 7 (Network Control and OAM).
- When using third-party vendor network equipment that has not been validated by Nortel, a pre-test of the network must be performed. This test includes mixed traffic going across the networks in different VLANs. The network specifications can meet the round trip delay and packet loss requirements.
- The round trip delay of the HSP VLAN must be less than 30 msec and the packet loss of the HSP VLAN must be below .1 % packet loss.
- The HSP port on the CP-PIV is set to auto-negotiate the link speed and duplex. Therefore, the network equipment that the CP-PIV is attached must also use auto-negotiate. Verify that both the CP-PIV and the network equipment speed and duplex are a match. The CP PII does not auto-negotiate; instead, it is fixed to 100 Mbps and full-duplex. Verify that both the CP PII and the network equipment speed and duplex are a match.
- Nortel recommends that MLT (Multi Link Trunking) be used across the enterprise IP network for the Campus Redundancy configuration.
- Cabling for the HSP port on the CP PII must be at least Cat 5 cabling. Cabling for the HSP port on the CP-PIV must be at least Cat 5e when running the link speed at 1 Gbps.

CAUTION
Duplex mismatches occur in the LAN environment when one side is set to Auto Negotiate and the other is hard configured.

The Auto Negotiate side adapts only to the speed setting of the fixed side. For duplex operations, the Auto Negotiate side sets itself to half-duplex mode. If the forced side is full-duplex, a duplex mismatch occurs.

Switching Equipment

Layer 2 switching equipment

The following equipment supports MLT (Multi Link Trunking), port based VLANs, and 802.1P priority configuration and is recommended for the HSP application.

- 325-24T - Layer 2 VLANs, MLT, 802.3ad
- 325-24G - Layer 2 VLANs, MLT, 802.3ad
- 425-24T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 425-48T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 460-24T-PWR - Layer 2 VLANs, MLT, DMLT, , 802.3ad, 802.3af PoE
- 470-24T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 470-48T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 5510-24T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 5510-48T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 5520-24T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing, 802.3af PoE
- 5520-48T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing, 802.3af PoE
- 8300 - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 8600 - Layer 2 VLANs, MLT, DMLT, SMLT, 802.3ad, L3 interVLAN routing

Third-party vendor switching equipment

The HSP supports all vendor switching equipment. The following third-party equipment has been tested:

- CISCO WS-3750G 24T-E GE ENH MULTILAYER CAYALYST (Layer 2 VLAN mode)
- 3C17203-3COM US/ 3COM 24-PORT 10/100TX SWITCH W/2
- 3COM 3C17304-US 3COM SS3 SWITCH 4228G 28PORTS EN
- 13240 EXTREME SUMMIT 200-24 SWITCH - 24 PORTS

ATTENTION

The HSP cannot be routed, and as a result, it cannot be extended through a Layer 3 router unless that device supports a method of providing Layer 2 end-to-end connectivity (Example: Layer 2 tunneling). Therefore, when passing through routing equipment, the HSP must remain in the same subnet from one Call Server to the other (Example: tunneling the HSP over the network).

HSP IP address configuration

The configuration of HSP IP addressing can be performed after the installation process if the default IP addresses are not appropriate for the customer network. Nortel strongly recommends allocation of a network IP address within a customer address space if the network is not dark fiber driven by BayStack470 switches.

CLASS network engineering rules

In a single-group network system, the network internal blocking is determined by the concentration ratio of equipped ports on Intelligent Peripheral Equipment and the number of interfaced loops or superloops. Depending on traffic engineering, a non-blocking network is achievable.

Feature operation

A call originated from Telephone A (or Trunk A) seeks to terminate on a CLASS Telephone B. When Telephone B starts to ring, Telephone A hears ringback. A unit in CLASS Modem (CMOD) is assigned to collect the originator's CND information and waits for the CND delivery interval. After the first ring at Telephone B, a silence period (deliver interval) ensues, and the CMOD unit begins to deliver CND information to the CLASS telephone.

The CND information of a traffic source (Telephone A) is a system information, which is obtained by the system when a call is originated. During the two-second ringing period of the CLASS Telephone B, Telephone A's CND is delivered to CMOD by SSD messages (using signaling channel only). When the CND information is sent from CMOD to CLASS Telephone B, it is delivered through a voice path during the four-second silence cycle of Telephone B. The CMOD unit is held for a duration of six seconds.

The system delivers SSD messages containing CND information to CMOD and then sends it to Telephone B during the delivery interval through a voice path.

Table 60 "CMOD Unit Capacity" (page 307) is the CMOD capacity table. It provides the number of CMOD units required to serve a given number of CLASS telephones with the desired GoS (P.001). The required number of CMOD units can have a capacity range whose upper limit is greater than the number of CLASS telephones equipped in a given configuration.

Table 60
CMOD Unit Capacity

CMOD Unit	CLASS Telephone	CMOD Unit	CLASS Telephone
1	1-2	33	2339-2436
2	3-7	34	2437-2535
3	8-27	35	2536-2635
4	28-59	36	2637-2735
5	60-100	37	2736-2835
6	101-150	38	2836-2936
7	151-206	39	2937-3037
8	207-267	40	3038-3139
9	268-332	41	3140-3241
10	333-401	42	3242-3344
11	402-473	43	3345-3447
12	474-548	44	3448-3550
13	549-625	45	3551-3653
14	626-704	46	3654-3757
15	705-785	47	3768-3861
16	786-868	48	3862-3966
17	869-953	49	3967-4070
18	954-1039	50	4071-4175
19	1040-1126	51	4176-4281
20	1127-1214	52	4282-4386
21	1215-1298	53	4387-4492
22	1299-1388	54	4493-4598
23	1389-1480	55	4599-4704
24	1481-1572	56	4705-4811
25	1573-1665	57	4812-4918
26	1666-1759	58	4919-5025
27	1760-1854	59	5026-5132

CMOD Unit	CLASS Telephone	CMOD Unit	CLASS Telephone
28	1855-1949	60	5133-5239
29	1950-2046	61	5240-5347
30	2047-2142	62	5348-5455
31	2143-2240	63	5456-5563
32	2241-2338	64	5564-5671

Guidelines for non-Call Center applications

In a non-call center application, there is no significant number of agent telephones. Therefore, no conversion of agent telephones to regular telephones is needed.

Engineering rule (no reconfiguration required)

The following engineering rule can be followed to avoid the need to reconfigure a switch to accommodate the CLASS feature: Provide the number of CMOD units serving all CLASS telephones in the system based on the capacity table (see Table 60 "CMOD Unit Capacity" (page 307)).

Guidelines for Call Center applications

Engineering rules (no reconfiguration required)

Follow these engineering rules to avoid the need to reconfigure a switch to accommodate the CLASS feature for a call center environment:

1. Convert agent telephones to regular telephones:

   1 agent CLASS telephone = 4 telephones (called equivalent telephones)

2. Sum up the total number of regular CLASS telephones and equivalent CLASS telephones and find the number of CMOD units required based on the capacity table (see Table 60 "CMOD Unit Capacity" (page 307)).

Configuration parameters

Design parameters are constraints on the system established by design decisions and enforced by software checks. Defaults are provided in the factory-installed database. However, some parameter values must be set manually, through the OA&M interface, to reflect the actual needs of the customer’s application.

For guidelines on how to determine appropriate parameter values for call registers, I/O buffers, and so on, see "Design parameters" (page 177) and "Memory engineering" (page 188).

Assigning loops and card slots in the CS 1000E

Contents

This section contains information on the following topics:

"Introduction" (page 309)

"Loops and superloops" (page 310)

"Card slot usage and requirements" (page 311)

"Assigning loops and cards in the CS 1000E" (page 314)

"Preparing the final card slot assignment plan" (page 322)

Introduction

Calculating the number and assignment of cards and, relatedly, Media Gateways is an iterative procedure, because of specific capacity and usage requirements.

In a CS 1000E system, Digital Signal Processor (DSP), Digitone receiver (DTR), Tone and Digit Switch (TDS), and other services are provided by circuit cards such as Media Cards, Media Gateway Controller (MGC) or the Small System Controller (SSC). These resources are available only to the MG 1000E (with optional Expander) that the circuit cards reside. Other services, such as Conference, are available as system resources but require MG 1000E-specific DSP resources in order to access them.

"System capacities" (page 187) and "Resource calculations" (page 215) describe the theoretical, traffic-based calculations used by EC to estimate the required number of Media Cards and MG 1000Es. This chapter describes the steps to allocate the cards to specific MG 1000Es. The process can result in an increase in the required number of Media Cards and MG 1000Es.

Note on terminology

Each MG 1000E can be connected to an optional MG 1000E Expander in order to increase the number of card slots available. In this chapter, the term *MG 1000E* includes the optional MG 1000E Expander, if equipped.

Loops and superloops

A fully expanded CS 1000E system provides a maximum of 256 loops or 64 superloops. Each superloop must be defined on a loop number that is a multiple of 4.

A superloop can be configured to include two MG 1000Es. In such a case, the first MG 1000E is referred to as shelf 0 and the second as shelf 1.

A maximum of 1024 TNs (= 2 × 16 × 32) from two MG 1000Es can be associated with a superloop.

Virtual superloops

There are no physical timeslots on MG 1000Es. Timeslots are defined within virtual superloops that benefit from the non-blocking timeslot architecture used by IP Phones and Virtual Trunks.

The superloop is layered into 16 banks of virtual superloops interfacing the 16 card slots in the two MG 1000Es. This expands the superloop's 120 timeslots to 1920 timeslots (= 16 × 120) to service a maximum of 1024 TNs in the address space. MG 1000Es are therefore non-blocking with respect to timeslots.

Internally, a card number separates the banks of software timeslots. Since a superloop is associated with 16 cards, each card is associated with one virtual superloop.

The network-level circuits, such as Conference and Tones, use additional loops outside of this address space. They also use DSPs from within the non-blocking superloops.

MGTDS and MGCONF loops

VXCT is replaced and decoupled in CS 1000 Release 5.0 with MGTDS and MGCONF. Media Gateway Tone and Conference configuration is configured in LD 17.

With MGTDS, you can configure two Media Gateway TDS loops. MGC-based MG 1000E lets 30 parties on each loop. SSC-based MG 1000E lets 16 parties on each loop.

With MGCONF, you can configure up to four Media Gateway Conference loops. MGC-based MG 1000E capacity is 2 MGCONF loops with 30 parties on each loop.

> **ATTENTION**
> Conference in a CS 1000E system with a mix of SSC and MGC as the controllers for the MG 1000Es:
>
> - If a conference loop is configured in LD 17 on any SSC-based MG 1000E, the maximum number of parties for all conferences on the system is 6.
> - If conference loops in LD 17 are only configured on MGC-based MG 1000Es, the maximum number of parties for all conferences on the system is 30.

PRI/PRI2/DTI/DTI2 loops

An MGC is required in slot 0 of any MG 1000E that will contain a PRI/PRI2/DTI/DTI2 card.

Each T1/E1 span consumes a loop, as well as a card slot.

The CP PII processor can support up to 40 PRI/PRI2/DTI/DTI2 spans.

The CP PIV and CP PM processors can support up to 100 PRI/PRI2/DTI/DTI2 spans. However, this many TI/E1 spans would consume most of the loops on the system.

Card slot usage and requirements

Table 61 "Card slots in the MG 1000E chassis" (page 311) summarizes the physical and logical card slots available in the MG 1000E chassis.

Table 62 "Card slots in the MG 1000E cabinet" (page 313) summarizes the physical and logical card slots available in the MG 1000E cabinet.

Table 61
Card slots in the MG 1000E chassis

Slot number		Used for	Comment
Physical	**Logical**		
MG 1000E chassis			
0	0	SSC or MGC	Dedicated card slot
If DTRs are configured in any other card slot, a receiver hardware pack must be equipped in the slot.			

Slot number		Used for	Comment
Physical	Logical		
1–4	1–4	• Media Cards • Digital line cards • Digital trunk cards (requires MGC) • Analog line cards • Analog trunk cards • Application cards	
n/a	5–6	n/a	Not supported
MG 1000E Expander			
7–10	7–10	• Media Cards • Digital line cards • Analog line cards • Analog trunk cards • Application cards	
Virtual			
n/a	0	32-port MGC DSP daughterboard	32-port DSP daugherboards use virtual slot 0. 32-port daugherboards are supported in MGC daugherboard location 1 and location 2.
n/a	11	96-port MGC DSP daughterboard	96-port DSP daughterboard uses virtual slots 11, 12, and 13. 96-port daugherboard is supported in MGC daughterboard location 1.
n/a	12		
n/a	13		
If DTRs are configured in any other card slot, a receiver hardware pack must be equipped in the slot.			

Slot number		Used for	Comment
Physical	**Logical**		
n/a	14	DTRs (maximum: 8)	Required if any analog terminals or trunks are equipped in the MG 1000E.
n/a	15	DTRs (maximum: 8)	
n/a	15	MF tone detectors (maximum: 4)	Must be provided on each MG 1000E that requires tone-based signaling.
If DTRs are configured in any other card slot, a receiver hardware pack must be equipped in the slot.			

Table 62
Card slots in the MG 1000E cabinet

Slot number		Used for	Comment
Physical	**Logical**		
MG 1000E cabinet			
0	0	SSC or MGC	Dedicated card slot
1–9	1–9	• Media Cards • Digital line cards • Digital trunk cards (requires MGC) • Analog line cards • Analog trunk cards • Application cards	
10	10	n/a	Not supported
Virtual			
n/a	0	32-port MGC DSP daughterboard	32-port DSP daugherboards use virtual slot 0. 32-port daugherboards are supported in MGC daugherboard location 1 and location 2.
If DTRs are configured in any other card slot, a receiver hardware pack must be equipped in the slot.			

Slot number		Used for	Comment
Physic al	Logical		
n/a	11	96-port MGC DSP daughterboard	96-port DSP daughterboard uses virtual slots 11, 12, and 13. 96-port daugherboard is supported in MGC daughterboard location 1.
n/a	12		
n/a	13		
n/a	14	DTRs (maximum: 8)	Required if any analog terminals or trunks are equipped in the MG 1000E.
n/a	15	DTRs (maximum: 8)	
n/a	15	MF tone detectors (maximum: 4)	Must be provided on each MG 1000E that requires tone-based signaling.
If DTRs are configured in any other card slot, a receiver hardware pack must be equipped in the slot.			

Assigning loops and cards in the CS 1000E

MG 1000Es are non-blocking with respect to timeslots (see "Virtual superloops" (page 310)). Blocking can occur only if an MG 1000E is configured with fewer DSP ports than the line or trunk ports require.

The following rules and guidelines describe methods to balance constraints and usage requirements. Use these guidelines to develop a detailed card slot and loop assignment plan.

Rules and guidelines

1. Place the MGC or SSC card in slot 0 of each MG 1000E.
2. There must be at least one Media Card in each MG 1000E.

> **ATTENTION**
>
> **IMPORTANT!**
>
> DSP resources cannot be shared between MG 1000Es. Therefore, each MG 1000E must contain sufficient Media Cards to provide the DSP resources required by the equipment configured in that MG 1000E.

3. There must be at least one TDS loop in each MG 1000E.
4. Allocate the users and Media Cards for dedicated DSPs first, then fill remaining empty slots in MG 1000Es with other IPE cards.

ATTENTION
There is no way to reserve DSP resources for dedicated usage (such as Conference). If a system has higher than expected call rates for standard telephones, these standard telephones can effectively hijack DSP resources required for dedicated functions. Therefore, in a system with high call rates for standard telephones, place dedicated and standard resources in different MG 1000Es.

Provision resources in the following order:

a. "Conference" (page 315)

b. "TDS" (page 316)

c. "Broadcast circuits" (page 317)

d. "Other service circuits" (page 319)

e. "TDM telephones and TDM agents" (page 319)

f. "Consoles" (page 319)

g. "Standard telephones" (page 320)

Conference

Each MG 1000E provides up to 64 conference circuits (ports), which can be used to form conferences of up to 30 parties each. Conferences are restricted to 6 parties each when using SSC. The MGC card has 60 conference circuits (2 loops). The SSC card has 32 conference circuits (2 loops), and another 32 conference circuits (2 loops) on the dual-port IP daughterboard. Users can configure 2 conference loops on each MGC-based MG 1000E, with each loop providing 30 conference circuits. Users can configure 4 conference loops on each SSC-based MG 1000E, with each loop providing 16 conference circuits.

The conference circuits are available to all MG 1000Es in the system. Calls are assigned to conference circuits on a "round robin" basis. Each conference circuit is accessed through a DSP port in the MG 1000E that the conference loop is defined. In addition, the device using the service can require another DSP in order to reach the conference port (see "DSP ports for Conference" (page 229)).

For non-blocking access, provide an equal number of DSP ports and conference ports. In other words, provide one 32-port Media Card for every pair of defined conference loops.

MG 1000E and Media Card calculations for Conference

1. Calculate the number of MG 1000Es required for Conference based on the number of conference circuits needed, in multiples of 60:

Number of MG 1000Es = ROUNDUP(Number of conference circuits ÷ 60)

2. Calculate the number of Media Cards required for Conference based on the number of conference circuits needed, in multiples of 32:

   Number of Media Cards = ROUNDUP(Number of conference circuits ÷ 32)

Examples

1. 30 conference circuits are needed:
   - One MG 1000E has Conference configured. All other MG 1000Es do not have conference circuits provisioned.
   - The MG 1000E with Conference requires one Media Card to support the service.

2. 33 conference circuits are needed:
   - One MG 1000E has Conference configured. All other MG 1000Es do not have conference circuits provisioned.
   - The MG 1000E with Conference requires two Media Cards to support the service.

3. 100 conference circuits are needed:
   - Two MG 1000Es have Conference configured. All other MG 1000Es do not have conference circuits provisioned.
   - Each MG 1000E with Conference requires two Media Cards to support the service.

TDS

A minimum of one TDS loop is required in each MG 1000E. The TDS circuits are provided by the MGC or SSC card. If additional TDS circuits are required in any MG 1000E, a second TDS loop can be configured in it.

When an MGTDS card is provisioned to provide additional conference ports, two additional TDS loops are automatically provisioned as well. However, these two TDS loops are disabled.

PRI/PRI2/DTI/DTI2

Each digital trunk in a CS 1000E system requires a dedicated DSP resource. Each T1 span requires 24 ports of DSP and each E1 span requires 30 ports of DSP.

PRI/PRI2/DTI/DTI2 cards require the use of CEMUX are supported in slots 1-4 of an MG 1000E chassis and slots 1-9 of a MG 1000E cabinet.

The definition of each PRI/PRI2/DTI/DTI2 span consumes 1 loop and can be configured in LD 17.

Controlled broadcast

If an MGate card is used for controlled broadcast, the rules for card placement of the MGate card and timeslot usage is the same as for a MiRan card. The MGate card will require 1 DSP for every listener.

Table 63 "Example of timeslot sharing in a superloop" (page 317), if timeslot sharing is used for MiRan, the same that would be used for MGate, when used for controlled broadcast.

Broadcast circuits

Music and Recorded Announcement (RAN) are broadcast circuits. One channel can support many listeners. Each listener needs one DSP port. A broadcast music trunk is required for every 60 broadcast users.

In order to maximize the number of simultaneous connections to a Nortel Integrated Recorded Announcer card in one MG 1000E shelf of a superloop, use all the timeslots for the superloop for that card. The software "steals" the timeslots from the other shelf of the superloop, provided the equivalent card slot in the second MG 1000E is not used. Table 63 "Example of timeslot sharing in a superloop" (page 317) illustrates the strategy.

Table 63
Example of timeslot sharing in a superloop

MG 1000E 0				MG 1000E 1			
l	**s**	**c**	**Card**	**l**	**s**	**c**	**Card**
0	0	1	Media Card for Conference	0	1	1	[Available for use.]
0	0	2	Media Card for Conference	0	1	2	[Available for use.]
0	0	3	Integrated Recorded Announcer card	0	1	3	[Must be left empty to avoid conflict with Integrated Recorded Announcer card.]
0	0	4	Media Card for RAN/Music	0	1	4	[Available for use.]
0	0	7	Media Card for RAN/Music	0	1	7	[Available for use.]
0	0	8	Media Card for RAN/Music	0	1	8	[Available for use.]
Legend: l = loop (superloop), s = shelf, c = card							

MG 1000E 0				MG 1000E 1			
l	**s**	**c**	**Card**	**l**	**s**	**c**	**Card**
0	0	9	Media Card for RAN/Music	0	1	9	[Available for use.]
0	0	10	[Leave empty — all DSPs in this MG 1000E are allocated to the conference and broadcast circuits, so there is no room for TDM devices.]	0	1	10	[Available for use.]
Legend: l = loop (superloop), s = shelf, c = card							

An alternative strategy is to use just one MG 1000E on a superloop when broadcast circuits are required.

Integrated Recorded Announcer card calculations

Since many listeners can be connected to a single source channel, it is possible that one or two sources on a broadcast source (Recorded Announcer) card can use all of the 120 timeslots available. No other sources can be used on that card.

Two Licenses are relevant for broadcast services:

- RAN CON = the maximum number of simultaneous RAN listeners in a system
- MUS CON = the maximum number of simultaneous music listeners in a system

When (RAN CON + MUS CON) > 120, more than one card is required for the music and RAN source. The following calculation provides the minimum number of cards required:

Number of cards = ROUNDUP[(RAN CON + MUS CON) ÷ 120]

For Recorded Announcer cards, the number of ports being ordered is known. Assuming that port usage and connection load is spread evenly across the Recorded Announcer cards, the calculation can be recast to perform the following check:

If [(RAN CON + MUS CON) ÷ Number of ports] <= 120, then there are sufficient Recorded Announcer cards to support the broadcast functions in the system.

MG 1000E DSP calculations for controlled broadcast

Since there are no ISMs to indicate when controlled broadcast is used, usage of this feature requires special attention.

- Symposium scripts must not assign more than 120 broadcasts to an MGate card.

- There must be enough DSPs in the MG1000E to support all of the functions within that chassis or cabinet. If 128 ports of call pilot are provided, 128 DSPs are required for call pilot usage. If controlled broadcast is used with 120 connections, then only 8 DSPs would be left for access to all remaining call pilot ports.

When too few DSPs are present and controlled broadcast is given, the end user would get a busy signal instead of the broadcast message.

MG 1000E and Media Card calculations for broadcast circuits

Since each Recorded Announcer card can broadcast to up to 120 listeners, each card requires a maximum of four 32-port Media Cards for all music or RAN source channels for that card. Each MG 1000E can support a single fully used music/RAN source card (1 broadcast source card + 4 Media Cards to support it).

- If (RAN CON + MUS CON) <= 32, allocate 1 Media Card and all Recorded Announcer cards to the same MG 1000E.
- If 32 < (RAN CON + MUS CON) <= 120, allocate 4 Media Cards + 1 Recorded Announcer card to the same MG 1000E.

Other service circuits

The list of other service circuits is extensive. It includes, amongst others, cards such as Nortel Integrated applications, CallPilot, and analog trunks.

Provide one DSP port for each channel of the service circuits.

TDM telephones and TDM agents

Provide one DSP port for each TDM telephone or TDM agent. Each XDLC card supports up to 16 TNs. (See also "Non-blocking access for ACD" (page 322).)

Consoles

Each M2250 attendant console and PC Console requires two TNs (originating and terminating) on an XDLC card and one Aux TN (for supervisor function). Nortel also recommends two power TNs per console.

DSPs are used when a call is active on an Attendant loop key. Each side (originating and terminating) requires one DSP, for a total of two DSPs per active/held call on the console.

Queued calls (ICI key indicators) do not consume DSP resources until the Attendant answers the call on a loop key.

DSP calculations

For standard access, provide 4 DSPs per console.

For dedicated DSPs, provide 12 DSPs per console (2 × 6 loop keys).

Standard telephones

Standard telephones are the average line users configured with a standard configuration.

1. Using a rule of thumb of five telephones per unallocated DSP, distribute line cards to the MG 1000Es with empty slots and unused DSPs.

   The rule of thumb is derived as follows:

   - A Media Card with 32 DSPs supports 794 CCS. This approximates to 24.8 CCS per DSP (794 ÷ 32).
   - The default value for average user traffic is 5 CCS. At 5 CCS per standard user, 24.8 CCS per DSP translates to 5 telephones per DSP.

2. Using a rule of thumb of one Media Card per seven line cards, fill empty MG 1000Es with the remaining line cards and their required Media Cards.

   The rule of thumb assumes average traffic of less than 7 CCS per telephone. This is derived as follows:

   - There are a total of 8 card slots available in each MG 1000E.
   - If 1 card slot is used by a Media Card, a maximum of 7 line cards, or 112 telephones (7 × 16 ports), can be added to the MG 1000E.
   - A Media Card with 32 DSPs supports 794 CCS. This is the traffic capacity of this particular MG 1000E.
   - A capacity limit of 794 CCS means each telephone must generate less than 7 CCS, on average (794 ÷ 112).

   For average traffic of more than 7 CCS per telephone, use Table 64 "Maximum number of Media Cards, line cards, and telephones in an MG 1000E" (page 320) to determine the number of Media Cards and telephones that can be assigned to an MG 1000E.

Table 64
Maximum number of Media Cards, line cards, and telephones in an MG 1000E

CCS per telephone	Media Cards	Line cards	Telephones*
<= 7.0	1	7	112
<= 8.0	1	6	96
*Number of telephones = Number of line cards × 16 ports			

CCS per telephone	Media Cards	Line cards	Telephones*
<= 10.0	1	5	80
<= 18.9	2	6	96
<= 22.8	2	5	80
<= 28.5	2	4	64
<= 36.0	3	5	80
*Number of telephones = Number of line cards × 16 ports			

3. Use a similar rule to add trunk cards (XUT) and their required Media Cards. See Table 65 "Maximum number of Media Cards, trunk cards, and trunks in an MG 1000E" (page 321).

Table 65
Maximum number of Media Cards, trunk cards, and trunks in an MG 1000E

CCS per telephone	Media Cards	Trunk cards	Trunks*
<= 14.2	1	7	56
<= 36.0	2	6	48
*Number of trunks = Number of trunk cards × 8 ports			

4. To mix line and trunk cards in an MG 1000E, calculate the total CCS for the number of lines and trunks. Then use Table 66 "Traffic capacity of Media Cards (Erlang B at P.01)" (page 321) to identify the number of Media Cards required to support that CCS rate.

Table 66
Traffic capacity of Media Cards (Erlang B at P.01)

Total CCS	Number of Media Cards	Number of DSPs
794	1	32
1822	2	64
2891	3	96

CLASS cards

CLASS cards can be placed in any MG 1000E. Therefore, each CLASS cards requires 32 ports of DSP.

The telephones that use the CLASS cards do require extra DSP resources. The rules for allocating standard telephones apply.

Non-blocking access for ACD

Table 67 "Number of Media Cards, line cards, and ACD agents per superloop" (page 322) describes two alternative recommended configurations to provide non-blocking ACD access to DSP ports. Typically, 33 CCS per agent telephone is engineered. However, in a non-blocking configuration, up to 36 CCS per agent is allowed.

Table 67
Number of Media Cards, line cards, and ACD agents per superloop

Total number of agents*	MG 1000E (shelf 0)			MG 1000E (shelf 1)		
	Media Cards	Line cards	Agents*	Media Cards	Line cards	Agents*
128	2	4	64	2	4	64
160	3	5	80	3	5	80
*Number of agents = Number of digital line cards × 16 ports; CCS per agent: 33–36						

Preparing the final card slot assignment plan

Prepare a final card slot assignment plan as follows:

1. Count the inventory of cards and MG 1000Es developed in accordance with this chapter.
2. Go back to the EC theoretical calculations and increment the order requirements to match the modified configuration.
3. Produce the final EC configuration report. EC output includes the following:
   - card locations
   - DSP allocations (where Media Card utilization is less than 100%)
   - MG 1000Es that must have conference circuits provisioned
   - MG 1000Es that must not have conference circuits provisioned

Provisioning

Contents

This section contains information on the following topics:

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"Step 1: Define and forecast growth" (page 324)

"Step 2: Estimate CCS per terminal" (page 325)

"Step 3: Calculate number of trunks required" (page 326)

"Step 4: Calculate line, trunk, and console load" (page 327)

"Step 5: Calculate Digitone receiver requirements" (page 328)

"Step 6: Calculate total system load" (page 329)

"Step 7: Calculate the number of IPE cards required" (page 329)

"Step 8: Calculate the number of Media Cards required" (page 329)

"Step 9: Calculate the number of Signaling Servers required" (page 329)

"Step 10: Provision conference/TDS loops" (page 329)

"Step 11: Calculate the number of Media Gateways required" (page 330)

"Step 12: Assign equipment and prepare equipment summary" (page 330)

"Media Card and Virtual Trunk worksheet" (page 330)

Introduction

This section provides a high-level overview of the steps required to determine general equipment requirements. Consult your Nortel representative and use a configuration tool, such as EC, to fully engineer a system.

> **ATTENTION**
>
> **IMPORTANT!**
>
> The values used in the examples in this chapter are for illustrative purposes only, and should not be interpreted as limits of the system capacity. The values must be adjusted to suit the application of a particular system.

Step 1: Define and forecast growth

Forecast the number of telephones required at two-year and five-year intervals.

The customer determines the number of telephones required when the system is placed in service (cutover). If the customer is unable to provide a two-year and five-year growth forecast, then use an estimate of annual personnel growth in percent to estimate the number of telephones required at the two-year and five-year intervals.

Example

A customer has 500 employees and needs 275 telephones to meet the system cutover. The customer projects an annual increase of 5% of employees based on future business expansion. The employee growth forecast is:

- 500 employees × 0.05 (percent growth) = 25 additional employees at 1 year
- 525 employees × 0.05 = 27 additional employees at 2 years
- 552 employees × 0.05 = 28 additional employees at 3 years
- 580 employees × 0.05 = 29 additional employees at 4 years
- 609 employees × 0.05 = 31 additional employees at 5 years
- 640 employees × 0.05 = 32 additional employees at 6 years

The ratio of telephones to employees is 275 ÷ 500 = 0.55.

To determine the number of telephones required from cutover through a five-year interval, multiply the number of employees required at each of the time periods by the ratio of telephones to employees (0.55).

- 500 employees × 0.55 = 275 telephones required at cutover
- 525 employees × 0.55 = 289 telephones required at 1 year

- 552 employees × 0.55 = 304 telephones required at 2 years
- 580 employees × 0.55 = 319 telephones required at 3 years
- 609 employees × 0.55 = 335 telephones required at 4 years
- 640 employees × 0.55 = 352 telephones required at 5 years

This customer requires 275 telephones at cutover, 304 telephones at two years, and 352 telephones at five years.

Each DN assigned to a telephone requires a TN. Determine the number of TNs required for each customer. Perform this calculation for cutover, two-year, and five-year intervals.

Step 2: Estimate CCS per terminal

Estimate the station and trunk centi-call seconds (CCS) per terminal (CCS/T) using any one of the following methods:

1. Comparative method
2. Manual calculation
3. Default method

Comparative method

Select three existing systems that have an historical record of traffic study data. The criteria for choosing comparative systems are:

1. Similar line size (+25%)
2. Similar business (such as bank, hospital, insurance, manufacturing)
3. Similar locality (urban or rural)

Calculate the average station, trunk, and intra-system CCS/T for the selected systems. Apply these averages to calculate trunk requirements for the system being provisioned.

Manual calculation

Normally, the customer can estimate the number of trunks required at cutover and specify the Grade-of-Service (GoS) to be maintained at two-year and five-year periods (see Table 68 "Example of manual calculation of CCS/T" (page 326)).

Use an appropriate trunking table (see Appendix "Reference tables" (page 335)) to obtain estimated trunk group usage for the number of trunks. Divide the number of lines that are accessing the group at cutover into the estimated usage. The result is the CCS/T, which can be used to estimate trunk requirements.

Table 68 "Example of manual calculation of CCS/T" (page 326) provides an example of the manual calculation.

Table 68
Example of manual calculation of CCS/T

Traffic source	Cutover (CCS)		Two years (CCS)		Five years (CCS)	
Line	275 × 6.2 =	1705	304 × 6.2 =	1885	352 × 6.2 =	2183
Trunk	275 × 4.1 =	1128	304 × 4.1 =	1247	352 × 4.1 =	1444
Subtotal		2833		3132		3627
Console		30		30		30
Total system load		2863		3162		3657
Line CCS/T = 6.2; Trunk CCS/T = 4.1; two consoles = 30 CCS.						

Repeat this method for each trunk group in the system, with the exception of small special services trunk groups (such as TIE, WATS, and FX trunks). Normally, customers tolerate a lesser GoS on these trunk groups.

Default method

Studies conducted estimate that the average line CCS/T is never greater than 5.5 in 90% of all businesses. If attempts to calculate the CCS/T using the comparative method or the manual calculation are not successful, the default of 5.5 line CCS/T can be used.

Determine the network line usage by multiplying the number of lines by 5.5 CCS/T. Then multiply the total by 2 to incorporate the trunk CCS/T. However, this method double-counts the intra-CCS/T, resulting in over-provisioning if the intra-CCS/T is high. Also, this method is not able to forecast individual trunk groups. The trunk and intra-CCS/T are forecast as a group total.

Step 3: Calculate number of trunks required

Once the trunk CCS/T is known and a GoS has been specified by the customer, determine the number of trunks required per trunk group to meet cutover, two-year, and five-year requirements. The following example demonstrates the method.

Example

The customer requires a Poisson 1% blocking GoS (see "Trunk traffic Poisson 1 percent blocking" (page 337)). The estimated trunk CCS/T is 1.14 for a DID trunk group. Determine the total trunk CCS by multiplying the number of lines by the trunk CCS/T for cutover, two-year, and five-year intervals:

Cutover	275 (lines) × 1.14 (trunk CCS/T) =	313.5 CCS
Two-year	304 (lines) × 1.14 (trunk CCS/T) =	346.56 CCS
Five-year	352 (lines) × 1.14 (trunk CCS/T) =	401.28 CCS

Use "Trunk traffic Poisson 1 percent blocking" (page 337) to determine the quantity of trunks required to meet the trunk CCS at cutover, two-year, and five-year intervals. In this case:

- 17 DID trunks are required at cutover
- 18 DID trunks are required in two years
- 21 DID trunk are required in five years

For trunk traffic greater than 4427 CCS, allow 29.5 CCS/T.

Step 4: Calculate line, trunk, and console load

CS 1000E

Line load

Calculate line load by multiplying the total number of TNs by the line CCS/T. The number of TNs is determined as follows:

- one TN for every DN assigned to one or more single-line telephones
- one TN for every multi-line telephone without data option
- two TNs for every multi-line telephone with data option

Trunk load

The number of Virtual Trunks to provision is calculated by the ordering and configuration tool as part of the Media Card provisioning calculation. See "Media Card and Virtual Trunk worksheet" (page 330) for the manual calculation.

Console load

Calculate console load by multiplying the number of consoles by 30 CCS per console.

Step 5: Calculate Digitone receiver requirements

Once station and trunk requirements have been determined for the complete system, calculate the Digitone receiver (DTR) requirements.

for information about the DTR resources provided by the MGC or SSC card and optional, additional XDTR cards, see "DTR" (page 199) .

In the CS 1000E, DTRs are not system-wide resources. They support only the telephones and trunks in the MG 1000E that they reside in. See "DTR" (page 199) for the calculations to determine overall DTR traffic and refer to reference tables "Digitone receiver requirements Model 1" (page 340) through "Digitone receiver requirements Model 4" (page 342) to estimate overall system requirements.

The actual provisioning of additional DTR resources depends on the number of Media Gateways in the system, and the distribution of line and trunk cards within them.

The models in reference tables "Digitone receiver requirements Model 1" (page 340) through "Digitone receiver requirements Model 4" (page 342) are based on some common PBX traffic measurements.

Model 1

"Digitone receiver requirements Model 1" (page 340) is based on the following factors:

- 33% intraoffice calls, 33% incoming calls, and 33% outgoing calls
- 1.5% dial tone delay GoS
- no Digitone DID trunks or incoming Digitone TIE trunks

Model 2

"Digitone receiver requirements Model 2" (page 341) is based on the following factors:

- the same traffic pattern as Model 1
- Digitone DID trunks or incoming Digitone TIE trunks
- Poisson 0.1% blockage GoS

Model 3

"Digitone receiver requirements Model 3" (page 341) is based on the following factors:

- 15% intraoffice calls, 28% incoming calls, and 56% outgoing calls
- 1.5% dial tone delay GoS
- no Digitone DID trunks or incoming Digitone TIE trunks

Model 4

"Digitone receiver requirements Model 4" (page 342) is based on the following factors:

- the same traffic pattern as Model 3
- Digitone DID trunks or incoming Digitone TIE trunks
- Poisson 0.1% blockage GoS

Step 6: Calculate total system load

Total the line, trunk, console, and DTR load for each customer to get the total load figure for cutover, two-year, and five-year intervals.

Step 7: Calculate the number of IPE cards required

Using the results of previous calculations for growth forecast and the number of DTRs, calculate the number of IPE cards required. Divide the number of digital telephone TNs, analog (500/2500-type) TNs, and trunk TNs by the number of TN assignments for each card. Round up each calculation to the next integer, then total the number of cards required.

Perform the calculations separately for cutover, two-year, and five-year intervals.

Step 8: Calculate the number of Media Cards required

"Media Card and Virtual Trunk worksheet" (page 330) provides a theoretical, traffic-based calculation for the number of Media Cards required in the system. This is the method followed by the ordering and configuration tool. The results provide a starting point for provisioning. Refer to "Assigning loops and card slots in the CS 1000E" (page 309) for additional rules and tips to distribute the Media Cards amongst the Media Gateways, in order to determine final Media Card requirements.

Step 9: Calculate the number of Signaling Servers required

The ordering and configuration tool calculates the number of Signaling Servers required. For a description of the calculation method, see "Signaling Server algorithm" (page 235) . For tips about avoiding over-provisioning, see "Manual adjustment of Signaling Server requirements" (page 264) .

Step 10: Provision conference/TDS loops

The MGC or SSC card provides conference/TDS functions. Refer to "Conference" (page 315) and "TDS" (page 316) for information on provisioning these functions within each MG 1000E.

Step 11: Calculate the number of Media Gateways required

Calculating the required number of Media Gateways is an iterative procedure, because certain resources must be provisioned within each MG 1000E. Refer to "Assigning loops and card slots in the CS 1000E" (page 309).

Step 12: Assign equipment and prepare equipment summary

The ordering and configuration tool produces a summary of the equipment requirements for the complete system at cutover. Assign the equipment. Adjust the equipment summary if necessary as a result of assignment procedures. Use the finalized equipment summary to order the equipment for the system.

> **ATTENTION**
> Another step you want to consider at this point is system security. For more information, refer to *Access Control Management Reference (NN43001-602)*.

Media Card and Virtual Trunk worksheet

This worksheet is in two parts:

- Worksheet 1a: Media Card calculation
- Worksheet 1b: Virtual Trunk calculation

Input constants	Input configuration data
R_I – intraoffice calls ratio	Number of analog telephones
R_T – tandem calls ratio	Number of digital telephones
I – incoming calls to total calls ratio	Number of IP Phones
O – outgoing calls to total calls ratio	Number of DECT telephones
P – IP calls to total calls ratio	Number of SIP Virtual Trunks (estimated)
V – Virtual Trunk calls to total trunk calls ratio	Number of H.323 Virtual Trunks (estimated)
v_S – SIP Virtual Trunk calls to total Virtual Trunk calls ratio	
v_H – H.323 Virtual Trunk calls to total Virtual Trunk calls ratio	

Input constants	Input configuration data
r_{CON} – Conference loop to traffic loop ratio (Default = 0.07) Hold time in seconds (AHT_{XX})for telephone to telephone, trunk to trunk, telephone to trunk, trunk to telephone	

Worksheet A
Media Card calculation

Item	Calculation formula
(1) TDM telephone CCS	= (Number of analog telephones + Number of digital telephones + Number of line-side T1/E1 ports) × _____ CCS/telephone
(2) IP telephone CCS	= [(Number of IP telephones – Number of IP ACD agents) × ______ CCS/IP telephone] + (Number of IP agent telephones × ______ CCS/agent)
(3) Total line CCS	= (1) + (2)
(4) TDM trunk CCS	= (Number of TDM trunks) × ____ CCS/trunk
(5) SIP Virtual Trunk CCS	= Number of SIP Virtual Trunks × ____ CCS/trunk
(6) H.323 Virtual Trunk CCS	= Number of H.323 Virtual Trunks × ____ CCS/trunk
(7) Total trunk CCS	= (4) + (5) + (6)
(8) Total system CCS (T_{CCS})	= (3) + (7)
(9) Weighted average holding time (WAHT)	= (R_I × AHT_{SS}) + (R_T × AHT_{TT}) + (I × AHT_{TS}) + (O × AHT_{ST})
(10) Total calls (T_{CALL})	= 0.5 × T_{CCS} × 100 ÷ WAHT
(11) Calls requiring DSP resources (C_{DSP})	= (a) + [2 × (b)] + (c) + [2 × (d)] + (e) + (f) + [2 × (g)] + (h) + (i) + [2 × (j)]
• (a) Intraoffice IP-TDM telephone calls	= T_{CALL} × R_I × 2 × P × (1 – P)
• (b) Intraoffice TDM-TDM telephone calls	= T_{CALL} × R_I × $(1 – P)^2$
• (c) Tandem VT-TDM trunk calls	= T_{CALL} × R_T × 2 × V × (1 – V)
• (d) Tandem TDM-TDM trunk calls	= T_{CALL} × R_T × $(1 – V)^2$
• (e) IP Phone-TDM trunk calls	= T_{CALL} × O × P × (1 – V)
• (f) TDM telephone-VT calls	= T_{CALL} × O × (1 – P) × V
• (g) TDM telephone-TDM trunk calls	= T_{CALL} × O × (1 – P) × (1 – V)
• (h) VT-TDM telephone calls	= T_{CALL} × I × V × (1 – P)
• (i) TDM trunk-IP telephone calls	= T_{CALL} × I × (1 – V) × P
• (j) TDM trunk-TDM telephone calls	= T_{CALL} × I × (1 – V) × (1 – P)

Item	Calculation formula
(12) DSP CCS (CCS_{DSP})	= C_{DSP} × WAHT ÷ 100
(13) Number of Media Cards required for general traffic	= CCS_{DSP} ÷ 794
(14) DSP channels for Conference	= (Total number of telephones × r_{CON} × 0.4) = (Total number of telephones × 0.028)
(15) DSP channels for applications	= (a) + (b) + (c) + (d) + (e) + (f) + (g)
• (a) CallPilot	= Number of CallPilot ports
• (b) Integrated Recorded Announcer	= Number of Integrated Recorded Announcer ports
• (c) Integrated Conference Bridge	= Number of Integrated Conference Bridge ports
• (d) Integrated Call Director	= Number of Integrated Call Director ports
• (e) Integrated Call Assistant	= Number of Integrated Call Assistant ports
• (f) Hospitality Integrated Voice Service	= Number of Hospitality Integrated Voice Service ports
• (g) Agent Greeting	= Number of Agent Greeting ports
(16) Total DSP channels	= [(13) × 32] + (14) + (15)
(17) Total Media Cards	= Roundup((16) ÷ 32)

Worksheet B
Virtual Trunk calculation

Call type	Calculation formula
(1) Virtual Trunk calls (C_{VT})	= (a) + (b) + (c) + (d) + (e) + (f)
• (a) Tandem VT-TDM trunk calls	= T_{CALL} × R_T × 2 × V × (1 – V)
• (b) IP-VT calls	= T_{CALL} × O × P × V
• (c) TDM telephone -VT calls	= T_{CALL} × O × (1 – P) × V
• (d) VT-TDM telephone calls	= T_{CALL} × I × V × (1 – P)
• (e) VT -IP Phone	= T_{CALL} × I × V × P
• (f) Tandem VT (H.323) to VT (SIP) calls	= T_{CALL} × R_T × V^2 × v_H × v_S × 2 × 2
(2) Virtual Trunk CCS (VT_{CCS})	= C_{VT} × WAHT ÷ 100
(3) SIP Virtual Trunk calls	= VT_{CCS} × v_S
(4) H.323 Virtual Trunk calls	= VT_{CCS} × v_H
If the calculated number of Virtual Trunks differs significantly from the original estimated number of Virtual Trunks (> 20%), Nortel recommends using the calculated Virtual Trunk number and repeating the calculation procedure to yield a more accurate number for required Media Cards and Virtual Trunks.	

Call type	Calculation formula
(5) Number of Virtual Trunks	= Roundup(VT_{CCS} ÷ 794 × 32)
(6) Virtual Trunk traffic in erlangs	= Roundup(VT_{CCS} ÷ 36) use this for LAN/WAN bandwidth calculation
If the calculated number of Virtual Trunks differs significantly from the original estimated number of Virtual Trunks (> 20%), Nortel recommends using the calculated Virtual Trunk number and repeating the calculation procedure to yield a more accurate number for required Media Cards and Virtual Trunks.	

Appendix A
Reference tables

List of tables

"Trunk traffic Erlang B with P.01 Grade-of-Service" (page 336)

Table 70 "Trunk traffic Poisson 1 percent blocking" (page 337)

Table 71 "Trunk traffic Poisson 2 percent blocking" (page 339)

"Digitone receiver requirements Model 1" (page 340)

"Digitone receiver requirements Model 2" (page 341)

"Digitone receiver requirements Model 3" (page 341)

"Digitone receiver requirements Model 4" (page 342)

"Digitone receiver load capacity 6 to 15 second holding time" (page 343)

"Digitone receiver load capacity 16 to 25 second holding time" (page 345)

"Digitione receiver requirement Poisson 0.1 percent blocking" (page 346)

"Conference and TDS loop requirements" (page 347)

"Digitone receiver provisioning" (page 348)

Trunk traffic Erlang B with P.01 Grade-of-Service

Table 69
Trunk traffic Erlang B (P.01)

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	21	462	41	1076	61	1724	81	2387
2	5.4	22	491	42	1108	62	1757	82	2419
3	16.6	23	521	43	1140	63	1789	83	2455
4	31.3	24	550	44	1171	64	1822	84	2488
5	49.0	25	580	45	1203	65	1854	85	2520
6	68.8	26	611	46	1236	66	1886	86	2552
7	90.0	27	641	47	1268	67	1922	87	2588
8	113	28	671	48	1300	68	1955	88	2621
9	136	29	702	49	1332	69	1987	89	2653
10	161	30	732	50	1364	70	2020	90	2689
11	186	31	763	51	1397	71	2052	91	2722
12	212	32	794	52	1429	72	2088	92	2758
13	238	33	825	53	1462	73	2120	93	2790
14	265	34	856	54	1494	74	2153	94	2822
15	292	35	887	55	1526	75	2185	95	2858
16	319	36	918	56	1559	76	2221	96	2891
17	347	37	950	57	1591	77	2254	97	2923
18	376	38	981	58	1624	78	2286	98	2959
19	404	39	1013	59	1656	79	2318	99	2992
20	433	40	1044	60	1688	80	2354	100	3028
101	3060	121	3740	141	4424	161	5119	181	5810
102	3092	122	3776	142	4460	162	5155	182	5843
103	3128	123	3809	143	4493	163	5188	183	5879
104	3161	124	3845	144	4529	164	5224	184	5915
105	3197	125	3877	145	4561	165	5260	185	5974
106	3229	126	3913	146	4597	166	5292	186	5983
107	3265	127	3946	147	4630	167	5328	187	6019
108	3298	128	3982	148	4666	168	5360	188	6052
109	3330	129	4014	149	4702	169	5396	189	6088
For trunk traffic greater than 6469 CCS, allow 32.35 CCS per trunk.									

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
110	3366	130	4050	150	4738	170	5429	190	6124
111	3398	131	4082	151	4770	171	5465	191	6156
112	3434	132	4118	152	4806	172	5501	192	6192
113	3467	133	4151	153	4842	173	5533	193	6228
114	3503	134	4187	154	4874	174	5569	194	6260
115	3535	135	4219	155	4910	175	5602	195	6296
116	3571	136	4255	156	4946	176	5638	196	6332
117	3604	137	4288	157	4979	177	5670	197	6365
118	3640	138	4324	158	5015	178	5706	198	6401
119	3672	139	4356	159	5051	179	5738	199	6433
120	3708	140	4392	160	5083	180	5774	200	6469
For trunk traffic greater than 6469 CCS, allow 32.35 CCS per trunk.									

Trunk traffic Poisson 1 percent blocking

Table 70
Trunk traffic Poisson 1 percent blocking

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	21	426	41	993	61	1595	81	2215
2	5.4	22	453	42	1023	62	1626	82	2247
3	15.7	23	480	43	1052	63	1657	83	2278
4	29.6	24	507	44	1082	64	1687	84	2310
5	46.1	25	535	45	1112	65	1718	85	2341
6	64	26	562	46	1142	66	1749	86	2373
7	84	27	590	47	1171	67	1780	87	2404
8	105	28	618	48	1201	68	1811	88	2436
9	126	29	647	49	1231	69	1842	89	2467
10	149	30	675	50	1261	70	1873	90	2499
11	172	31	703	51	1291	71	1904	91	2530
12	195	32	732	52	1322	72	1935	92	2563
13	220	33	760	53	1352	73	1966	93	2594
14	244	34	789	54	1382	74	1997	94	2625
For trunk traffic greater than 6068 CCS, allow 30.34 CCS per trunk.									

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
15	269	35	818	55	1412	75	2028	95	2657
16	294	36	847	56	1443	76	2059	96	2689
17	320	37	876	57	1473	77	2091	97	2721
18	346	38	905	58	1504	78	2122	98	2752
19	373	39	935	59	1534	79	2153	99	2784
20	399	40	964	60	1565	80	2184	100	2816
101	2847	121	3488	141	4134	161	4786	181	5442
102	2879	122	3520	142	4167	162	4819	182	5475
103	2910	123	3552	143	4199	163	4851	183	5508
104	2942	124	3594	144	4231	164	4884	184	5541
105	2974	125	3616	145	4264	165	4917	185	5574
106	3006	126	3648	146	4297	166	4549	186	5606
107	3038	127	3681	147	4329	167	4982	187	5639
108	3070	128	3713	148	4362	168	5015	188	5672
109	3102	129	3746	149	4395	169	5048	189	5705
110	3135	130	3778	150	4427	170	5081	190	5738
111	3166	131	3810	151	4460	171	5114	191	5771
112	3198	132	3843	152	4492	172	5146	192	5804
113	3230	133	3875	153	4525	173	5179	193	5837
114	3262	134	3907	154	4557	174	5212	194	5871
115	3294	135	3939	155	4590	175	5245	195	5904
116	3326	136	3972	156	4622	176	5277	196	5937
117	3359	137	4004	157	4655	177	5310	197	5969
118	3391	138	4037	158	4686	178	5343	198	6002
119	3424	139	4070	159	4721	179	5376	199	6035
120	3456	140	4102	160	4754	180	5409	200	6068
For trunk traffic greater than 6068 CCS, allow 30.34 CCS per trunk.									

Trunk traffic Poisson 2 percent blocking

Table 71
Trunk traffic Poisson 2 percent blocking

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	31	744	61	1659	91	2611	121	3581
2	7.9	32	773	62	1690	92	2643	122	3614
3	20.9	33	803	63	1722	93	2674	123	3647
4	36.7	34	832	64	1752	94	2706	124	3679
5	55.8	35	862	65	1784	95	2739	125	3712
6	76.0	36	892	66	1816	96	2771	126	3745
7	96.8	37	922	67	1847	97	2803	127	3777
8	119	38	952	68	1878	98	2838	128	3810
9	142	39	982	69	1910	99	2868	129	3843
10	166	40	1012	70	1941	100	2900	130	3875
11	191	41	1042	71	1973	101	2931	131	3910
12	216	42	1072	72	2004	102	2964	132	3941
13	241	43	1103	73	2036	103	2996	133	3974
14	267	44	1133	74	2067	104	3029	134	4007
15	293	45	1164	75	2099	105	3051	135	4039
16	320	46	1194	76	2130	106	3094	136	4072
17	347	47	1225	77	2162	107	3126	137	4105
18	374	48	1255	78	2194	108	3158	138	4138
19	401	49	1286	79	2226	109	3190	139	4171
20	429	50	1317	80	2258	110	3223	140	4204
21	458	51	1348	81	2290	111	3255	141	4237
22	486	52	1374	82	2322	112	3288	142	4269
23	514	53	1352	83	2354	113	3321	143	4302
24	542	54	1441	84	2386	114	3353	144	4335
25	571	55	1472	85	2418	115	3386	145	4368
26	562	56	1503	86	2450	116	3418	146	4401
27	627	57	1534	87	2482	117	3451	147	4434
28	656	58	1565	88	2514	118	3483	148	4467
For trunk traffic greater than 4533 CCS, allow 30.2 CCS per trunk.									

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
29	685	59	1596	89	2546	119	3516	149	4500
30	715	60	1627	90	2578	120	3548	150	4533
For trunk traffic greater than 4533 CCS, allow 30.2 CCS per trunk.									

Digitone receiver requirements Model 1

Table 72
Digitone receiver requirements Model 1

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
2	7	2	17	1181	319
3	33	9	18	1244	336
4	69	19	19	1348	364
5	120	33	20	1455	393
6	179	49	21	1555	420
7	249	68	22	1662	449
8	332	88	23	1774	479
9	399	109	24	1885	509
10	479	131	25	1988	537
11	564	154	26	2100	567
12	659	178	27	2211	597
13	751	203	28	2325	628
14	848	229	29	2440	659
15	944	255	30	2555	690
16	1044	282			
See Step 5: Calculate Digitone receiver requirements for Model 1 assumptions.					

Digitone receiver requirements Model 2

Table 73
Digitone receiver requirements Model 2

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
2	2	2	17	843	253
3	21	7	18	920	276
4	52	15	19	996	299
5	90	27	20	1076	323
6	134	40	21	1153	346
7	183	55	22	1233	370
8	235	71	23	1316	395
9	293	88	24	1396	419
10	353	107	25	1480	444
11	416	126	26	1563	469
12	483	145	27	1650	495
13	553	166	28	1733	520
14	623	187	29	1816	545
15	693	208	30	1903	571
16	770	231			
See Step 5: Calculate Digitone receiver requirements" for Model 2 assumptions.					

Digitone receiver requirements Model 3

Table 74
Digitone receiver requirements Model 3

Number of DTRs	Max. number of Digitone	DTR load (CCS)	Number of DTRs	Max. number of Digitone	DTR load (CCS)

	lines			lines	
2	5	2	17	862	319
3	22	9	18	908	336
4	50	19	19	983	364
5	87	33	20	1062	393
6	132	49	21	1135	420
7	180	68	22	1213	449
8	234	88	23	1294	479
9	291	109	24	1375	509
10	353	131	25	1451	537
11	415	154	26	1532	567
12	481	178	27	1613	597
13	548	203	28	1697	628
14	618	229	29	1781	659
15	689	255	30	1864	690
16	762	282			
See Step 5: Calculate Digitone receiver requirements" for Model 3 assumptions.					

Digitone receiver requirements Model 4

Table 75
Digitone receiver requirements Model 4

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
2	4	2	17	683	253
3	18	7	18	745	276
4	41	15	19	808	299
5	72	27	20	872	323
6	109	40	21	935	346
7	148	55	22	1000	370
8	193	71	23	1067	395
9	240	88	24	1132	419
10	291	107	25	1200	444
11	340	126	26	1267	469
See Step 5: Calculate Digitone receiver requirements" for Model 4 assumptions.					

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
12	391	145	27	1337	495
13	448	166	28	1405	520
14	505	187	29	1472	545
15	562	208	30	1543	571
16	624	231			
See Step 5: Calculate Digitone receiver requirements" for Model 4 assumptions.					

Digitone receiver load capacity 6 to 15 second holding time

Table 76
Digitone receiver load capacity 6 to 15 second holding time

Numb er of DTRs	Average holding time in seconds									
	6	7	8	9	10	11	12	13	14	15
1	0	0	0	0	0	0	0	0	0	0
2	3	2	2	2	2	2	2	2	2	2
3	11	10	10	9	9	9	9	8	8	8
4	24	23	22	21	20	19	19	19	18	18
5	41	39	37	36	35	34	33	33	32	32
6	61	57	55	53	52	50	49	49	48	47
7	83	78	75	73	71	69	68	67	66	65
8	106	101	97	94	91	89	88	86	85	84
9	131	125	120	116	113	111	109	107	106	104
10	157	150	144	140	136	133	131	129	127	126
11	185	176	170	165	161	157	154	152	150	148
12	212	203	196	190	185	182	178	176	173	171
13	241	231	223	216	211	207	203	200	198	196
14	270	259	250	243	237	233	229	225	223	220
15	300	288	278	271	264	259	255	251	248	245
16	339	317	307	298	292	286	282	278	274	271
17	361	346	335	327	320	313	310	306	302	298
18	391	377	365	356	348	342	336	331	327	324
Load capacity is measured in CCS.										

Number of DTRs	Average holding time in seconds									
	6	7	8	9	10	11	12	13	14	15
19	422	409	396	386	378	371	364	359	355	351
20	454	438	425	414	405	398	393	388	383	379
21	487	469	455	444	435	427	420	415	410	406
22	517	501	487	475	466	456	449	443	438	434
23	550	531	516	504	494	487	479	472	467	462
24	583	563	547	535	524	515	509	502	497	491
25	615	595	579	566	555	545	537	532	526	521
26	647	628	612	598	586	576	567	560	554	548
27	680	659	642	628	618	607	597	589	583	577
28	714	691	674	659	647	638	628	620	613	607
29	746	724	706	690	678	667	659	651	644	637
30	779	758	738	723	709	698	690	682	674	668
31	813	792	771	755	742	729	719	710	703	696
32	847	822	805	788	774	761	750	741	733	726
33	882	855	835	818	804	793	781	772	763	756
34	913	889	868	850	836	825	812	803	795	787
35	947	923	900	883	867	855	844	835	826	818
36	981	957	934	916	900	886	876	866	857	850
37	1016	989	967	949	933	919	909	898	889	881
38	1051	1022	1001	982	966	951	938	928	918	912
39	1083	1055	1035	1015	999	984	970	959	949	941
40	1117	1089	1066	1046	1029	1017	1002	990	981	972
Load capacity is measured in CCS.										

Digitone receiver load capacity 16 to 25 second holding time

Table 77
Digitone receiver load capacity 16 to 25 second holding time

Number of DTRs	Average holding time in seconds									
	16	17	18	19	20	21	22	23	24	25
1	0	0	0	0	0	0	0	0	0	0
2	2	2	2	2	2	2	2	2	2	2
3	8	8	8	8	8	8	8	8	8	8
4	18	18	18	18	18	17	17	17	17	17
5	31	31	31	30	30	30	30	30	30	29
6	47	46	46	45	45	45	45	44	44	44
7	64	63	63	62	62	62	61	61	61	60
8	83	82	82	81	80	80	79	79	79	78
9	103	102	101	100	100	99	99	98	98	97
10	125	123	122	121	121	120	119	119	118	118
11	147	145	144	143	142	141	140	140	139	138
12	170	168	167	166	165	164	163	162	161	160
13	193	192	190	189	188	186	185	184	184	183
14	218	216	214	213	211	210	209	208	207	206
15	243	241	239	237	236	234	233	232	231	230
16	268	266	264	262	260	259	257	256	255	254
17	294	292	290	288	286	284	283	281	280	279
18	322	319	317	314	312	311	309	308	306	305
19	347	344	342	339	337	335	334	332	331	329
20	374	371	368	366	364	361	360	358	356	355
21	402	399	396	393	391	388	386	385	383	381
22	431	427	424	421	419	416	414	412	410	409
23	458	454	451	448	445	442	440	438	436	434
24	486	482	478	475	472	470	467	465	463	461
25	514	510	506	503	500	497	495	492	490	488
26	544	539	535	532	529	526	523	521	518	516
27	573	569	565	561	558	555	552	549	547	545
28	603	598	594	590	587	584	581	578	576	573
29	631	626	622	618	614	611	608	605	602	600
Load capacity is measured in CCS.										

Number of DTRs	Average holding time in seconds									
	16	17	18	19	20	21	22	23	24	25
30	660	655	651	646	643	639	636	633	631	628
31	690	685	680	676	672	668	665	662	659	656
32	720	715	710	705	701	698	694	691	688	686
33	751	745	740	735	731	727	724	721	718	715
34	782	776	771	766	761	757	754	750	747	744
35	813	807	801	796	792	788	784	780	777	774
36	841	835	829	824	820	818	814	810	807	804
37	872	865	859	854	849	845	841	837	834	831
38	902	896	890	884	879	875	871	867	863	860
39	934	927	921	914	909	905	901	897	893	890
40	965	958	952	945	940	936	931	927	923	920
Load capacity is measured in CCS.										

Digitione receiver requirement Poisson 0.1 percent blocking

Table 78
Digitone receiver requirements Poisson 0.1 percent blocking

Number of DTRs	DTR load (CCS)	Number of DTRs	DTR load (CCS)
1	0	26	469
2	2	27	495
3	7	28	520
4	15	29	545
5	27	30	571
6	40	31	597
7	55	32	624
8	71	33	650
9	88	34	676
10	107	35	703
11	126	36	729
12	145	37	756
13	166	38	783
14	187	39	810
15	208	40	837

Number of DTRs	DTR load (CCS)	Number of DTRs	DTR load (CCS)
16	231	41	865
17	253	42	892
18	276	43	919
19	299	44	947
20	323	45	975
21	346	46	1003
22	370	47	1030
23	395	48	1058
24	419	49	1086
25	444	50	1115

Conference and TDS loop requirements

Table 79
Conference and TDS loop requirements

Network loops required at 2 years	TDS loops required	Conference loops required
1–12	1	1
13–24	2	2
25–36	3	3
37–48	4	4
49–60	5	5
61–72	6	6
73–84	7	7
85–96	8	8
97–108	9	9
109–120	10	10

Digitone receiver provisioning

Table 80
Digitone receiver provisioning

DTR CCS	DTR ports	DTR CCS	DTR ports
1–2	2	488–515	24
3–9	3	516–545	25
10–19	4	546–576	26
20–34	5	577–607	27
35–50	6	608–638	28
51–69	7	639–667	29
70–89	8	668–698	30
90–111	9	699–729	31
112–133	10	730–761	32
134–157	11	762–793	33
158–182	12	794–825	34
183–207	13	826–856	35
208–233	14	857–887	36
234–259	15	888–919	37
260–286	16	920–951	38
287–313	17	952–984	39
314–342	18	985–1017	40
343–371	19	1018–1050	41
372–398	20	1051–1084	42
399–427	21	1085–1118	43
428–456	22	1119–1153	44
457–487	23	1154–1188	45
1189–1223	46	1961–1995	68
1224–1258	47	1996–2030	69
1259–1293	48	2031–2065	70
1294–1329	49	2066–2100	71
1330–1365	50	2101–2135	72
1366–1400	51	2136–2170	73
1401–1435	52	2171–2205	74
1436–1470	53	2206–2240	75
Provisioning assumes an 11-second holding time.			

DTR CCS	DTR ports	DTR CCS	DTR ports
1471–1505	54	2241–2275	76
1506–1540	55	2276–2310	77
1541–1575	56	2311–2345	78
1576–1610	57	2346–2380	79
1611–1645	58	2381–2415	80
1646–1680	59	2416–2450	81
1681–1715	60	2451–2485	82
1716–1750	61	2486–2520	83
1751–1785	62	2521–2555	84
1786–1802	63	2556–2590	85
1821–1855	64	2591–2625	86
1856–1890	65	2626–2660	87
1891–1926	66	2661–2695	88
1926–1960	67	2696–2730	89
2731–2765	90	2941–2975	96
2766–2800	91	2976–3010	97
2801–2835	92	3011–3045	98
2836–2870	93	3046–3080	99
2871–2905	94	3081–3115	100
2906–2940	95	3116–3465	101
Provisioning assumes an 11-second holding time.			

Nortel Communication Server 1000

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