Communication Server 1000E Planning and Engineering
Avaya Communication Server 1000

Release 7.6
NN43041-220
Issue 06.01
March 2013

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Contents

Chapter 1: New in this release

The following section details what is new in this document for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.6.

Navigation

Features

There are no new features introduced with this release.

Other changes

This section contains the following topic:

Revision history

March 2013	Standard 06.01. This document is up-issued to support Avaya Communication Server 1000 Release 7.6.
August 2012	Standard 05.13. This document is up-issued to include the section IPv6 for Linux servers.
May 2012	Standard 05.12. This document is up-issued to include additional information on the Media Application Server (MAS) in the Resource calculations and Application Engineering sections.
April 2012	Standard 05.11. This document is up-issued to include information about the surge-suppression cable for certain trunk cards.
March 2012	Standard 05.10. This document is up-issued to include updates to CSQI/ CSQO limits.

March 2012	Standard 05.09. This document is up-issued to include information about supported codecs and the MAS session connection call rate.
February 2012	Standard 05.08. This document is up-issued for changes in technical content. The section Signaling Server capacity limits on page 234 is updated.
November 2011	Standard 05.07. This document is up-issued to include an update to the number of IP Attendant Consoles available for IP Media Services sessions.
July 2011	Standard 05.06. This document is up-issued to update the Security Server capacities table with MAS cph details.
June 2011	Standard 05.05. This document is up-issued to include the Avaya Common Server (HP DL360 G7).
March 2011	Standard 05.04. This document is up-issued to include recommended fax configurations in the Communication Server 1000 Release 7.5.
February 2011	Standard 05.03. This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.
November 2010	Standard 05.02. This document is issued to support Avaya Communication Server 1000 Release 7.5.
November 2010	Standard 05.01. This document is issued to support Avaya Communication Server 1000 Release 7.5.
August 2012	Standard 04.06. This document is up-issued to include the section IPv6 for Linux servers.
March 2012	Standard 04.05. This document is up-issued to include updates to CSQI/ CSQO limits.
March 2012	Standard 04.04. This document is up-issued to include information about supported codecs and the MAS session connection call rate.
October 2010	Standard 04.03. This document is up-issued to update the dedicated Signaling Server capacity limits to support Avaya Communication Server 1000 Release 7.0.
August 2010	Standard 04.02. This document is up-issued to update planning and engineering capacities, and the Signaling Server algorithm to support Avaya Communication Server 1000 Release 7.0.
June 2010	Standard 04.01. This document is issued to support Avaya Communication Server 1000 Release 7.0.
February 2010	Standard 03.08. This document is up-issued to replace Baystack with Ethernet Routing Switch.
October 2009	Standard 03.07. This document is up-issued to include Media Gateway Extended Peripheral Equipment Controller (MG XPEC) content.
September 2009	Standard 03.06. This document is up-issued to include Media Gateway 1010 content.

August 2009	Standard 03.05. This document is up-issued to update Memory-Related parameters.
June 2009	Standard 03.04. This document is up-issued to include an introduction to the CP PM Co-resident Call Server and Signaling server feature in the New in this release chapter, and to add SIP in the Data network planning for VoIP chapter.
June 2009	Standard 03.03. This document is up-issued to include updates for Signaling Server algorithm constants.
May 2009	Standard 03.02. This document is up-issued to include a task flow graphic for Communication Server 1000 Release 6.0.
May 2009	Standard 03.01. This document is issued to support Nortel Communication Server 1000 Release 6.0.
October 2008	Standard 02.07. This document is up-issued to include information about Call Servers memory capacity.
October 2008	Standard 02.06. This document is up-issued to include the limit of 128 ITG cards or 4096 DSPs as a system limit. The table System Parameters includes the updated technical content.
May 2008	Standard 02.05. This document is up-issued to reflect changes in technical content for mobile extensions.
May 2008	Standard 02.04. This document is up-issued to include information about the maximum call rate supported in the Signaling Server software element.
February 2008	Standard 02.03. This document is up-issued to include additional Modem Pass Through Allowed information, corrections to software degradation release factors, and COTS servers running NRS on Linux cannot run signaling server applications.
January 2008	Standard 02.02. This document is up-issued to include Modem Pass Through Allowed information, updates for supported IP Phones, and Enterprise Configurator.
December 2007	Standard 02.01. This document is issued to support Nortel Communication Server 1000 Release 5.5.
December 2007	Standard 01.05. This document is up-issued with corrections to the memory requirements and number of IP users in PD/CL/RL applications. This document also contains corrections to CP PM supported slots in an Option 11C cabinet.
November 2007	Standard 01.04. This document is up-issued to include Survivable and Distributed Media Gateway ELAN traffic estimations.
July 2007	Standard 01.03. This document is up-issued with corrections to ISP 1100 Signaling Server Network Routing Service limits for calls per hour, end points, and routing entries.
June 2007	Standard 01.02. This document is up-issued with corrections to real time factors and software release degradation factors. This document also

	contains revised maximum number of IP users in PD/CL/RL applications.
May 2007	Standard 01.01. This document is issued to support Nortel Communication Server 1000 Release 5.0. This document is renamed Communication Server 1000E Planning and Engineering (NN43041-220) and contains information previously contained in the following legacy document, now retired: Communications Server 1000E: Planning and Engineering (553-3041-120).

Chapter 2: Customer service

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Navigation

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Getting technical support from the Avaya Web site

The easiest and most effective way to get technical support for Avaya products is from the Avaya Technical Support Web site at www.avaya.com/support.

Chapter 3: Overview of the engineering process

Contents

This chapter contains the following topics:

Introduction

Warning:

Before an Avaya Communication Server 1000E (Avaya 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 about the minimum VoIP network requirements and converging a data network with VoIP, see *Avaya 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 Avaya 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.

About this document

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

Subject

⚠ **Warning:**

Before an Avaya 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 about the minimum VoIP network requirements and converging a data network with VoIP, see *Avaya 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 Enterprise Configurator provides an alternative to the manual processes given in this document. It is beyond the scope of this document to describe the Enterprise Configurator process.

Note on legacy products and releases

This document contains information about systems, components, and features that are compatible with CS 1000 software. For more information about legacy products and releases, click the **Technical Documentation** link under **Support & Training** on the Avaya home page:

http://www.avaya.com

Applicable systems

This document applies to the 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 Avaya 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 Avaya PBX systems.

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

Conventions

In this document, CS 1000E is referred to generically as system.

In this document, the following Chassis or Cabinets are referred to generically as Media Gateway:

- Option 11C Mini Chassis (NTDK91) and Chassis Expander (NTDK92)
- Option 11C Cabinet (NTAK11)
- Avaya MG 1000E Chassis (NTDU14) and Expansion Chassis (NTDU15)
- Media Gateway 1010 (MG 1010) (NTC310)
- IPE module (NT8D37) with MG XPEC card (NTDW20)

In this document, the following cards are referred to generically as Gateway Controller:

- Media Gateway Controller (MGC) card (NTDW60 and NTDW98)
- Common Processor Media Gateway (CP MG) card (NTDW56 and NTDW59)
- Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card (NTDW20)

In this document, the following hardware platforms are referred to generically as Server:

- Call Processor Pentium IV (CP PIV)
- Common Processor Pentium Mobile (CP PM)
- Common Processor Media Gateway (CP MG)
- Common Processor Dual Core (CP DC)
- Commercial off-the-shelf (COTS) servers
    - IBM x306m server (COTS1)
    - HP DL320 G4 server (COTS1)
    - IBM x3350 server (COTS2)
    - Dell R300 server (COTS2)
    - HP DL360 G7 (Avaya Common Server)

In this document, the generic term COTS refers to all COTS servers. The term COTS1, COTS2, or Common Server refers to the specific servers in the preceding list.

Co-res CS and SS is not supported on COTS1 servers (IBM x306m, HP DL320-G4)

The following table shows CS 1000 supported roles for various hardware platforms.

Table 1: Hardware platform supported roles

Hardware platform	VxWorks Server	Linux Server	Co-res CS and SS	Gateway Controller
CP PIV	yes	no	no	no
CP PM	yes	yes	yes	no
CP DC	no	yes	yes	no
CP MG	no	no	yes (see note)	yes (see note)
MGC	no	no	no	yes
MG XPEC	no	no	no	yes
COTS1	no	yes	no	no
COTS2	no	yes	yes	no
Common Server	no	yes	yes	no

Note:

The CP MG card functions as the Co-res CS and SS, and the Gateway Controller while occupying slot 0 in a Media Gateway.

Related information

This section lists information sources that relate to this document.

Documents

The following documents are referenced in this document:

- *Avaya Feature Listing Reference, NN43001-111*
- *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*
- *Avaya Network Routing Service Fundamentals, NN43001-130*
- *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*
- *Avaya Electronic Switched Network Signaling and Transmission Guidelines, NN43001-280*
- *Avaya Transmission Parameters, NN43001-282*
- *Avaya Dialing Plans Reference, NN43001-283*
- *Avaya Circuit Card Reference, NN43001-311*
- *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*
- *Avaya Branch Office Installation and Commissioning, NN43001-314*
- *Avaya Linux Platform Base and Applications Installation and Commissioning, NN43001-315*
- *Avaya SIP Line Fundamentals, NN43001-508*
- *Avaya Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*
- *Avaya Automatic Call Distribution Fundamentals, NN43001-551*
- *Avaya System Management Reference, NN43001-600*
- *Avaya Access Control Management Reference, NN43001-602*
- *Avaya Software Input Output Administration, NN43001-611*
- *Avaya Security Management, NN43001-604*
- *Avaya Element Manager System Reference - Administration, NN43001-632*
- *Avaya Telephones and Consoles Fundamentals, NN43001-567*
- *Avaya IP Phones Fundamentals (NN43001-368), NN43001-368*
- *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*
- *Avaya Basic Network Feature Fundamentals, NN43001-579*
- *Avaya ISDN Basic Rate Interface Feature Fundamentals, NN43001-580*
- *Avaya Traffic Measurement Formats and Outputs Reference, NN43001-750*
- *Avaya Software Input Output Reference Maintenance, NN43001-711*

- *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering, NN43021-220*
- *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*
- *Avaya Communication Server 1000E Software Upgrades, NN43041-458*
- *Avaya CallPilot Planning and Engineering, 555-7101-101*

Online

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Engineering a new system

Figure 1: Engineering a new system on page 27 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, Avaya 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.

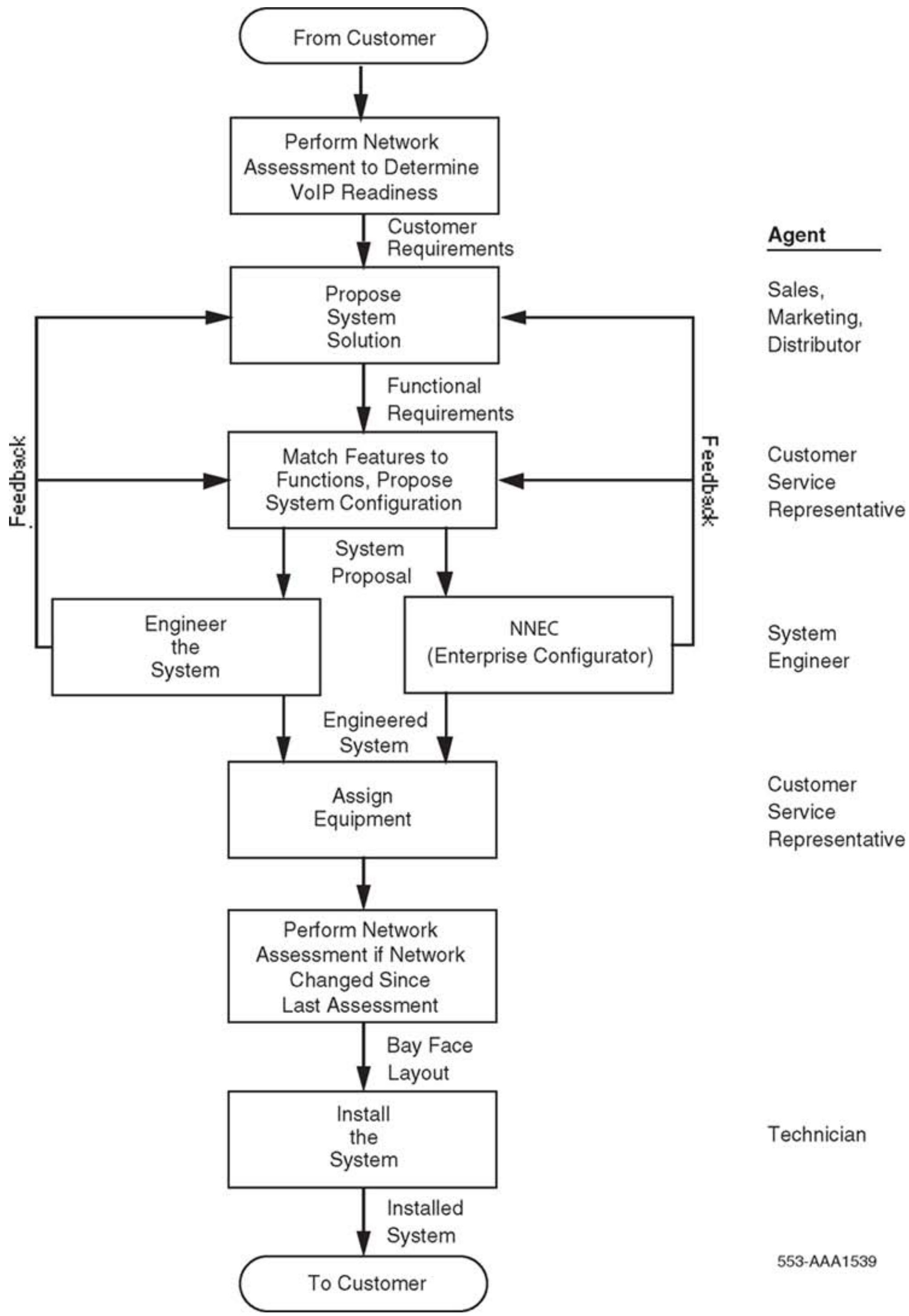

Figure 1: Engineering a new system

Communication Server 1000 task flow

This section provides high-level task flow diagrams for CS 1000 network and CS 1000 SA/HA system installations or upgrades. The task flow indicates the recommended sequence of events to follow when configuring a system and provides the document number that contains the detailed procedures required for the task. For more information refer to the following publications, which are referenced in the task flow diagram:

- *Avaya Linux Platform Base and Applications Installation and Commissioning, NN43001-315*
- *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*
- *Avaya Communication Server 1000E Software Upgrades, NN43041-458*

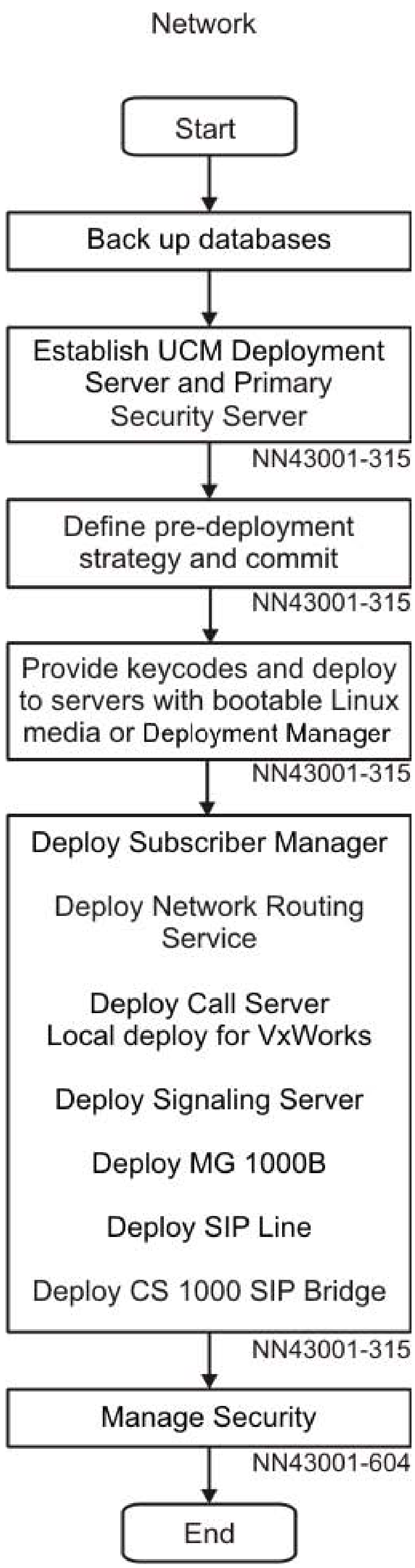

Figure 2: Communication Server 1000E network task flow

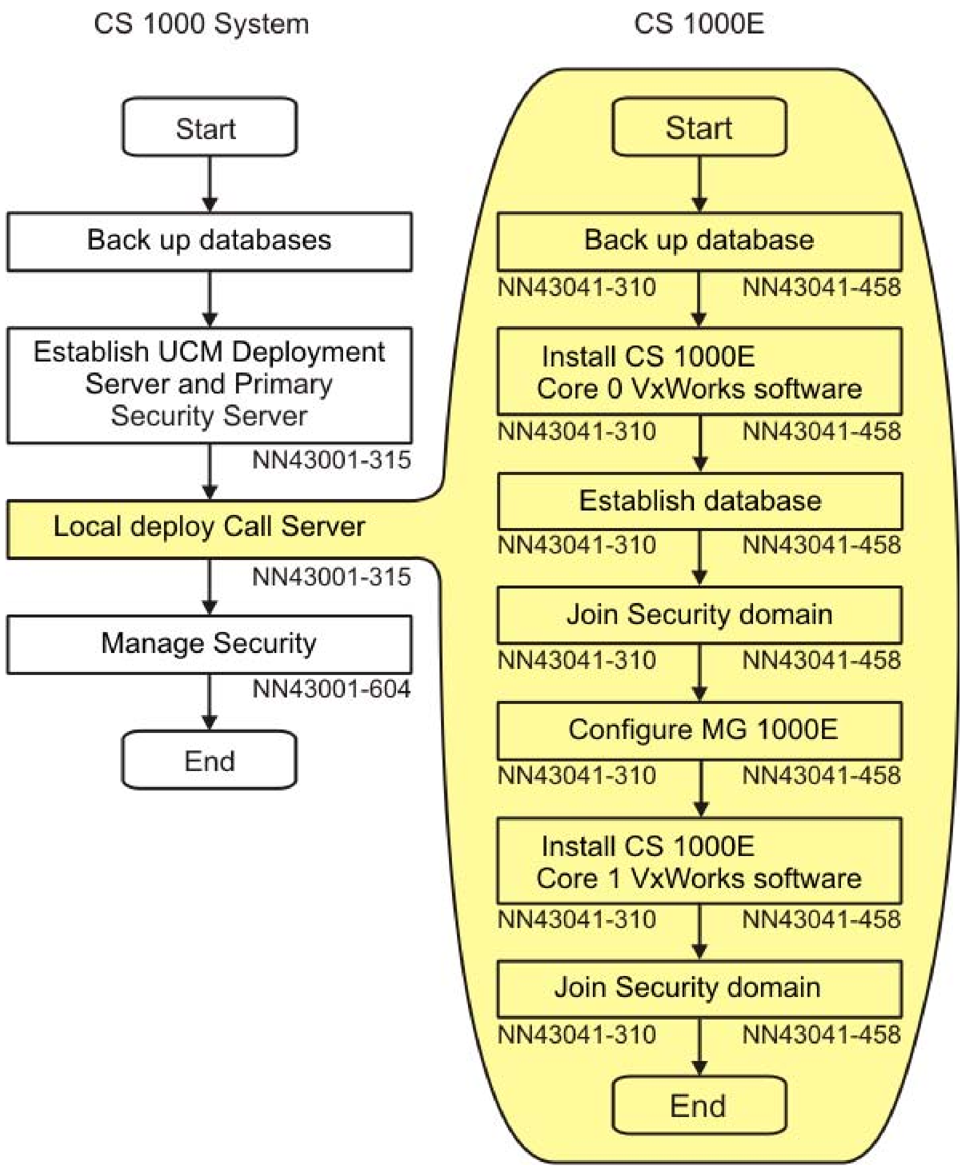

Figure 3: Communication Server 1000E deployment task flow

Enterprise Configurator

The Enterprise Configurator (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 Avaya web site. Enterprise Configurator 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.

Chapter 4: Regulatory information

Contents

This chapter contains the following topics:

System approval

The Avaya Communication Server 1000E (Avaya 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 Avaya CS 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 CS 1000M, CS 1000S, and CS 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 2: EMC specifications for Class A devices on page 34 lists the EMC specifications for the system.

Table 2: 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 Avaya. 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.

Voltage:

DANGER OF ELECTRIC SHOCK

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

Voltage:

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 3: Network connection specifications on page 38 contains information that must be given to the local telephone company when ordering standard network interface jacks for the system.

Table 3: Network connection specifications on page 38 includes columns for system port identification, Facility Interface Code (FIC), Service Order Code (SOC), Uniform Service Order Code (USOC) jack identification, and associated Avaya equipment part numbers.

Table 3: 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	02GS2	9.0F	2.7A	RJ21X	NT8D14

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
(2-Wire, Local Switched Access, Ground-Start)				CA21X*	
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
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 Avaya. 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 45° C (113° 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 45° C (113° 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
Avaya 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.

Voltage:

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 Avaya 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 Avaya 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.

Voltage:

WARNUNG

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

Chapter 5: Data network planning for VoIP

Contents

This chapter contains the following topics:

Introduction

⚠ **Warning:**

Before an Avaya Communication Server 1000E (Avaya 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 about the minimum VoIP network requirements and converging a data network with VoIP, see *Avaya 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 Avaya CS 1000E system.

See *Avaya 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 about 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 *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*:

- VoIP technology
  - H.323 protocols
  - Session Initiation Protocol (SIP)
  - VoIP concepts and protocols
  - Real-time Transport Protocol (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 *Avaya 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 *Avaya 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 *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260* for information about 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.

Chapter 6: System architecture

Contents

This chapter contains the following topics:

Main components and architecture

A typical Avaya Communication Server 1000E (Avaya CS 1000E) solution is composed of a Call Server, Media Gateway, and Signaling Server. You can also add components such as Layer 2 switches and optional Terminal Servers.

- The Call Server provides call processing capability. You can deploy an Avaya CS 1000E Standard Availability, High Availability, or Co-resident Call Server and Signaling Server. (see Communication Server 1000E Call Server on page 56)
- The Media Gateway cabinet or chassis platform provides the CS 1000E system with IPE card slots for connection of telephones and trunks. The Media Gateway houses the Gateway Controller and can house Server cards. Call Servers can support up to 50 Media Gateways and optional Avaya CS 1000 Media Gateway 1000E (Avaya MG 1000E) expander chassis. (see Media Gateway on page 66).

- 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 on page 230)
- 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 on page 91).
- The Terminal Server is an option that provides the CS 1000E system with additional serial ports for applications and maintenance. For more information about the MRV Terminal Server , see Terminal Server on page 90.

CS 1000E systems can be configured for either Standard Availability, High Availability (system redundancy), or Co-resident Call Server and Signaling Server (Co-res CS and SS).

Figure 4: CP PM Standard Availability on page 53 shows the typical main components of a Standard Availability CP PM Communication Server 1000E solution.

Figure 5: Co-resident Call Server and Signaling Server on page 54 shows the typical main components of a Co-resident Call Server and Signaling Server CS 1000E solution.

Figure 6: CP PM High Availability on page 55shows the typical main components of a High Availability CP PM equipped CS 1000E solution.

Figure 7: CP PIV High Availability on page 56 shows the typical main components of a High Availability CP PIV equipped CS 1000E solution.

Important:

CP PIV equipped CS 1000E cannot be configured for Standard Availability. CS 1000E systems equipped with a stand-alone 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. CS 1000E systems with Co-resident Call Server and Signaling Server cannot be configured for High Availability. The remainder of this chapter discusses each component in further detail.

The Communication Server 1000E system supports various types of hardware platforms. You must ensure that your hardware platform can support your target CS 1000E configuration. For more information about supported roles for each hardware platform, see Table 1: Hardware platform supported roles on page 24

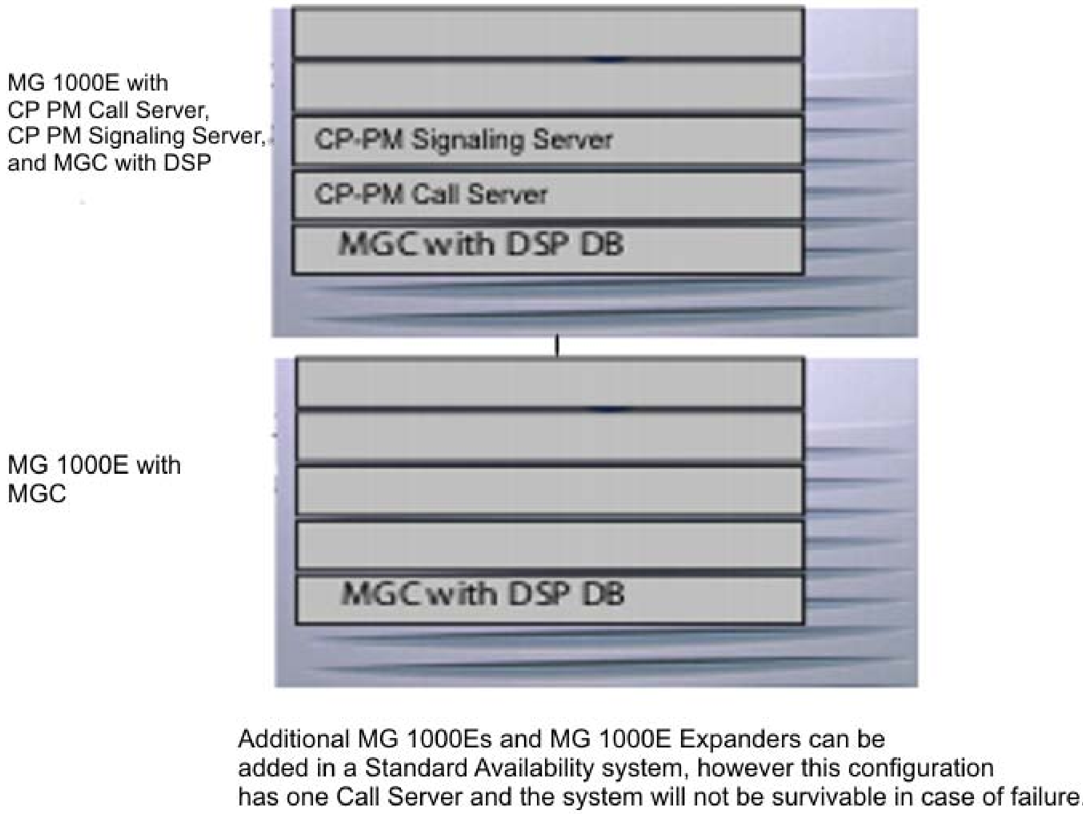

Figure 4: CP PM Standard Availability

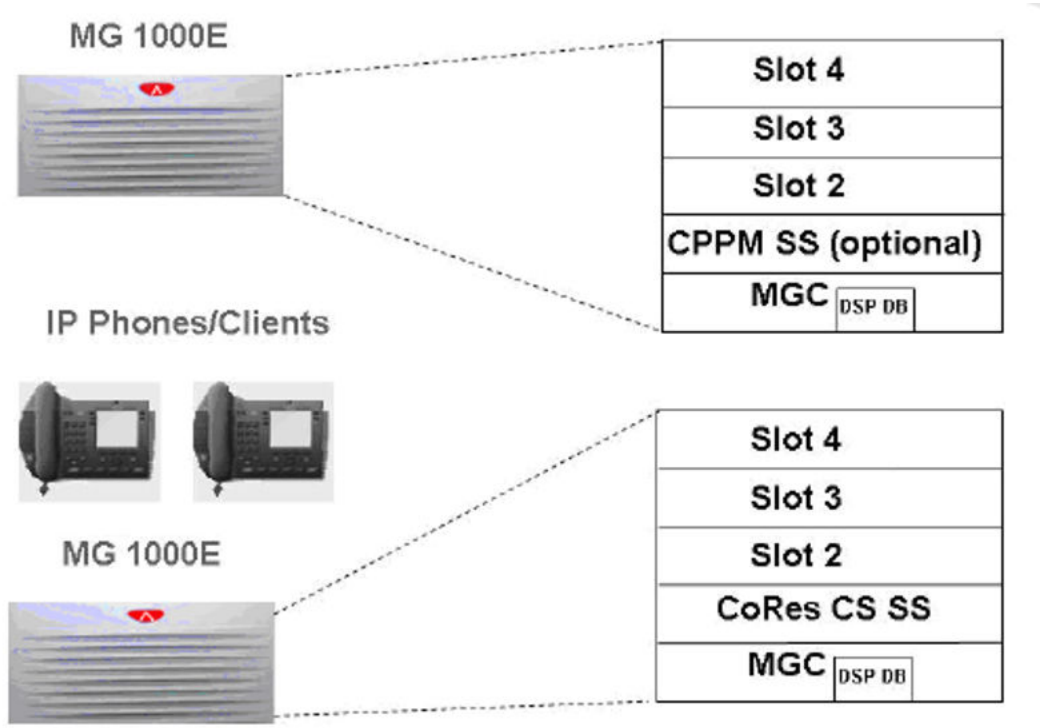

Figure 5: Co-resident Call Server and Signaling Server

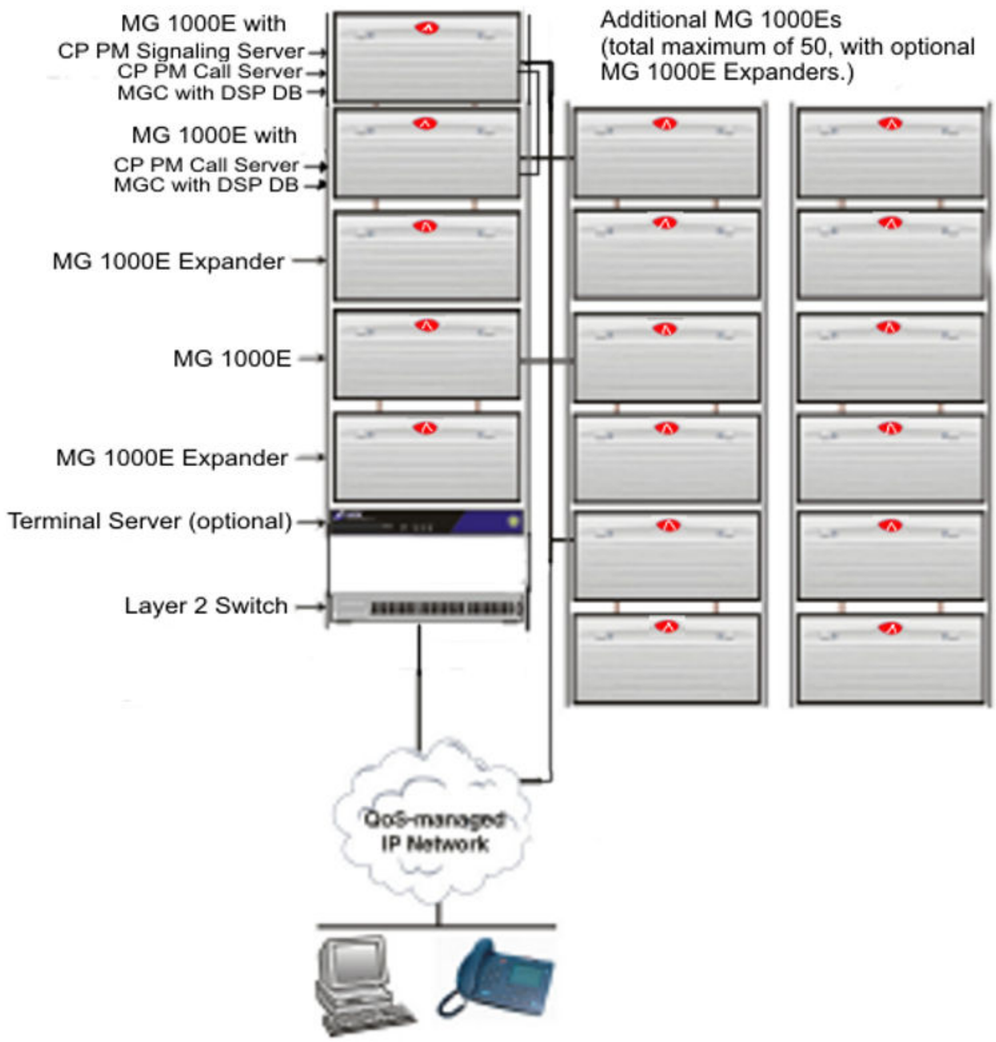

Figure 6: CP PM High Availability

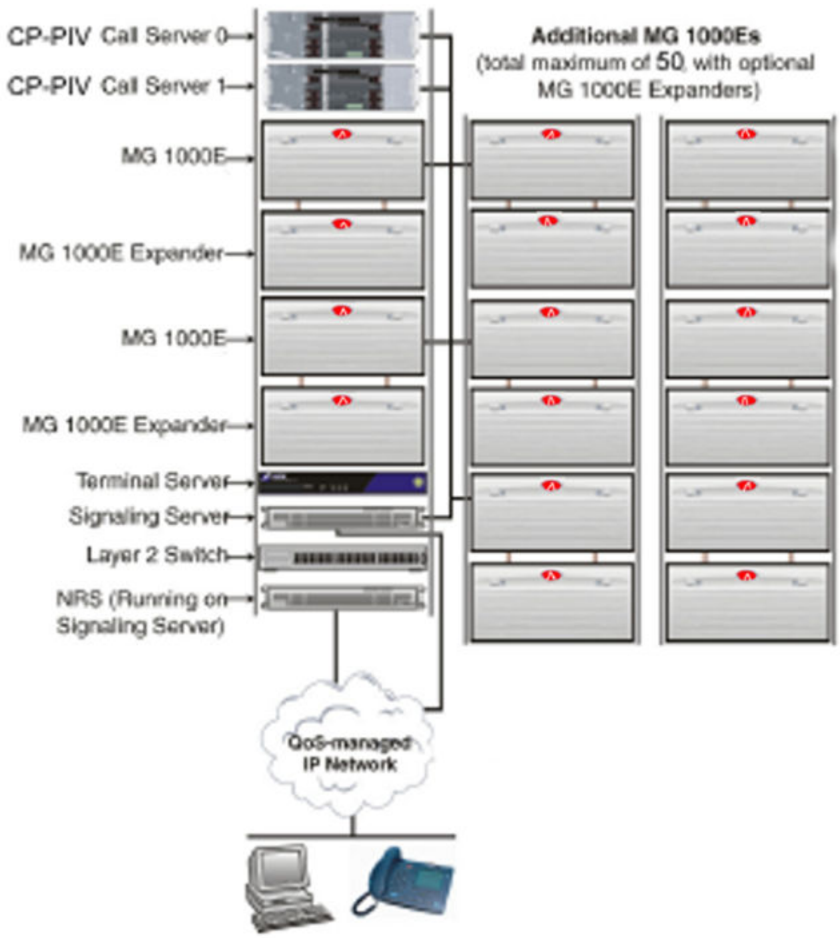

Figure 7: CP PIV High Availability

Communication Server 1000E Call Server

The CS 1000E Call Server serves as the call processor for the CS 1000E system. You can deploy a Call Server in a stand-alone Standard Availability (SA), stand-alone High Availability (HA) or in a Co-resident Call Server and Signaling Server (Co-res CS and SS) configuration. You can add another CP PM to a stand-alone SA system to create a CP PM equipped HA system (redundant system).

The CS 1000E SA and HA system software is VxWorks based. The CS 1000E Co-res CS and SS software is Linux based.

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 Media Gateways
- provide resources for system administration and user database maintenance

Operating parameters

The CS 1000E can be equipped as SA (single Call Server) or High Availability (dual Call Server) (Core 0 and Core 1) to provide a fully redundant system. The CP PIV supports High Availability only. The Co-resident Call Server and Signaling Server does not support High Availability.

Core 0 and Core 1 can operate in redundant mode over the High Speed Pipe (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 Media Gateways. This proprietary protocol is similar to industry-standard Media Gateway Control Protocol (MGCP) or H.248 Gateways.

CS 1000E Call Servers can control up to 50 Media Gateways.

The Call Servers provide connectivity to telephony devices using IP signaling through Media Gateways 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 *Avaya Circuit Card Reference, NN43001-311*

Physical description of a CP PIV Call Server

Figure 8: CS 1000E CP PIV Call Server (front and rear) on page 58 shows the front (without cover) and rear of one Call Server.

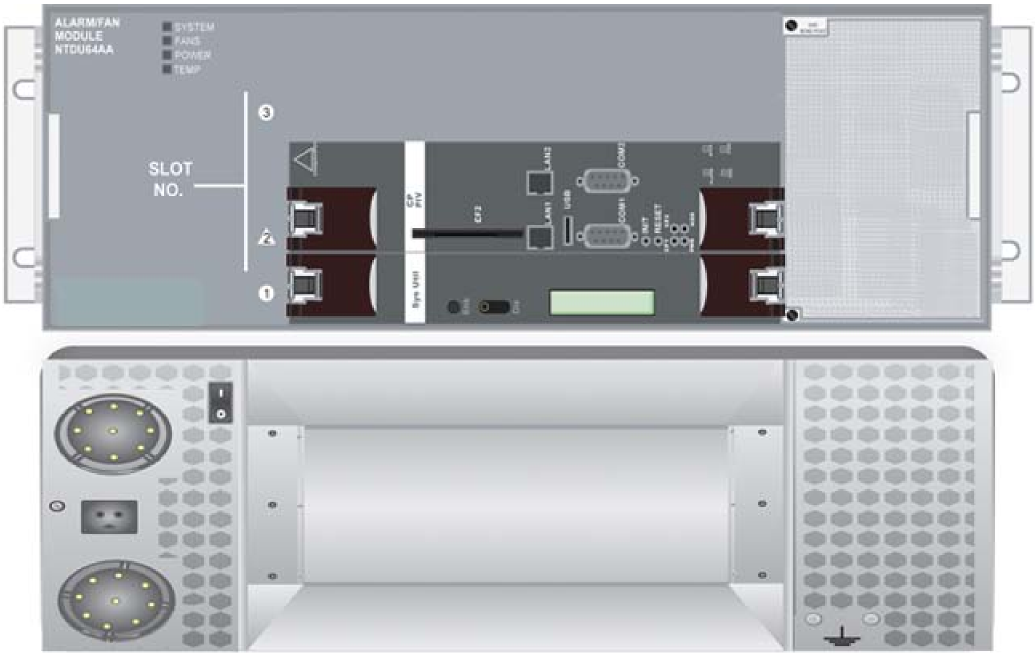

Figure 8: CS 1000E CP PIV Call Server (front and rear)

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 is an RS232 serial port you directly connect to a system terminal for system access. You can optionally connect the Com 1 port to an IP-based Terminal

Server, which provides standard serial ports for system maintenance and third-party applications (for more information, see Terminal Server on page 90).

- The Com 2 port is 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 Media Gateways. The port is a 10/100/1000MB autonegotiate 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 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 9: CS 1000E Media Gateway with CP PM Call Server (front) on page 61 shows the front (without cover) of a NTDU14 Media Gateway 1000E chassis. For a CP PM Call Server configuration, the Media Gateway contains a Media Gateway Controller (MGC) card and Call Processor Pentium Mobile (CP PM) card.

Figure 10: Option 11C cabinet upgraded to CS 1000E CP PM Call Server on page 62 shows the front (without cover) of an Option 11C cabinet upgraded with MGC and CP PM cards.

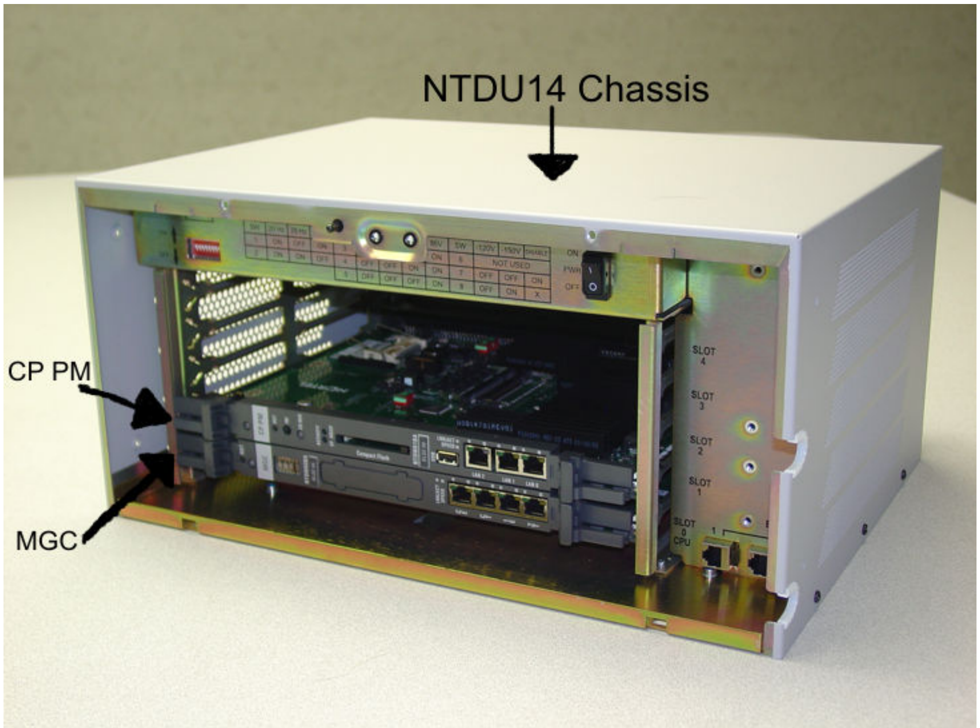

Figure 9: CS 1000E Media Gateway with CP PM Call Server (front)

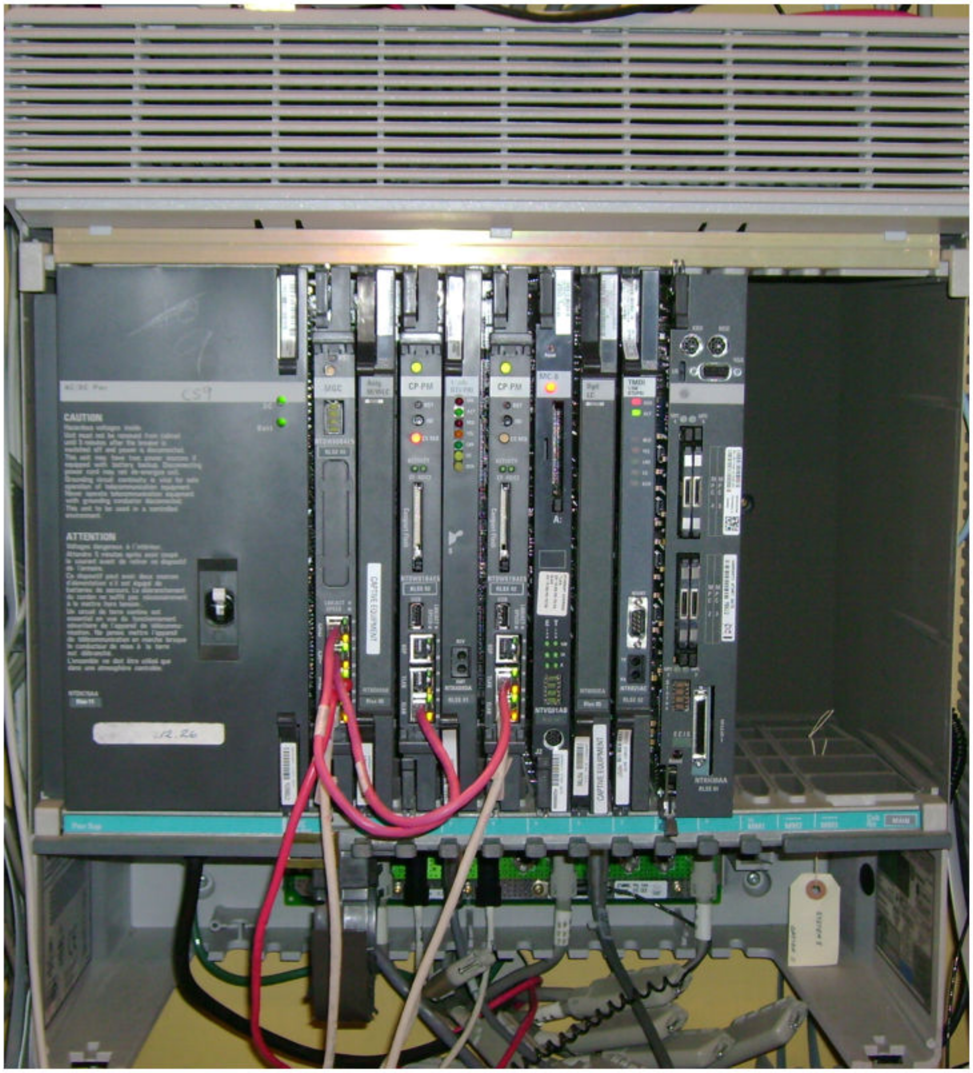

Figure 10: Option 11C cabinet upgraded to CS 1000E CP PM Call Server

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)

- Option 11C Mini chassis (except for slot 0 and slot 4)
- Option 11C Mini expander chassis
- Avaya Media Gateway 1000E main chassis (except for slot 0)
- Avaya Media Gateway 1000E expander chassis
- Media Gateway 1010 (except for slot 0)
- IPE module with MG XPEC card (except for controller slot)

• CP PM card

• MGC card with DSP daughterboards in each Media Gateway chassis or cabinet

• MG XPEC card in each IPE module

CP PM chassis

The CP PM is a circuit card that you insert in a Media Gateway. For more information, see Media Gateway on page 66 . For information about upgrading Option 11C equipment to support the CP PM card, see *Avaya Communication Server 1000E Upgrades, NN43041-458*.

For information about upgrading CS 1000M IPE modules to support CS 1000E Server cards, see *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering, NN43021-220* and *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

CP PM card

The Common Processor Pentium Mobile (CP PM) card can be configured as a Call Server. The CP PM offers similar features to that of the CP PIV processor, but uses an IPE slot form factor, allowing for a CS 1000E product with only a single Media Gateway chassis. The CP PM card can also be configured as a stand-alone Signaling Server, or a Co-resident Call Server and Signaling Server.

Figure 11: CP PM card NTDW61 on page 64 shows the CP PM faceplate and CP PM circuit card. The NTDW61 CP PM card is designed for use in Media Gateway IPE slots.

The NTDW99 CP PM card contains a metal faceplate that provides enhanced EMC containment. The NTDW99 CP PM card is designed for use in Media Gateway 1010 chassis slots 22 and 23.

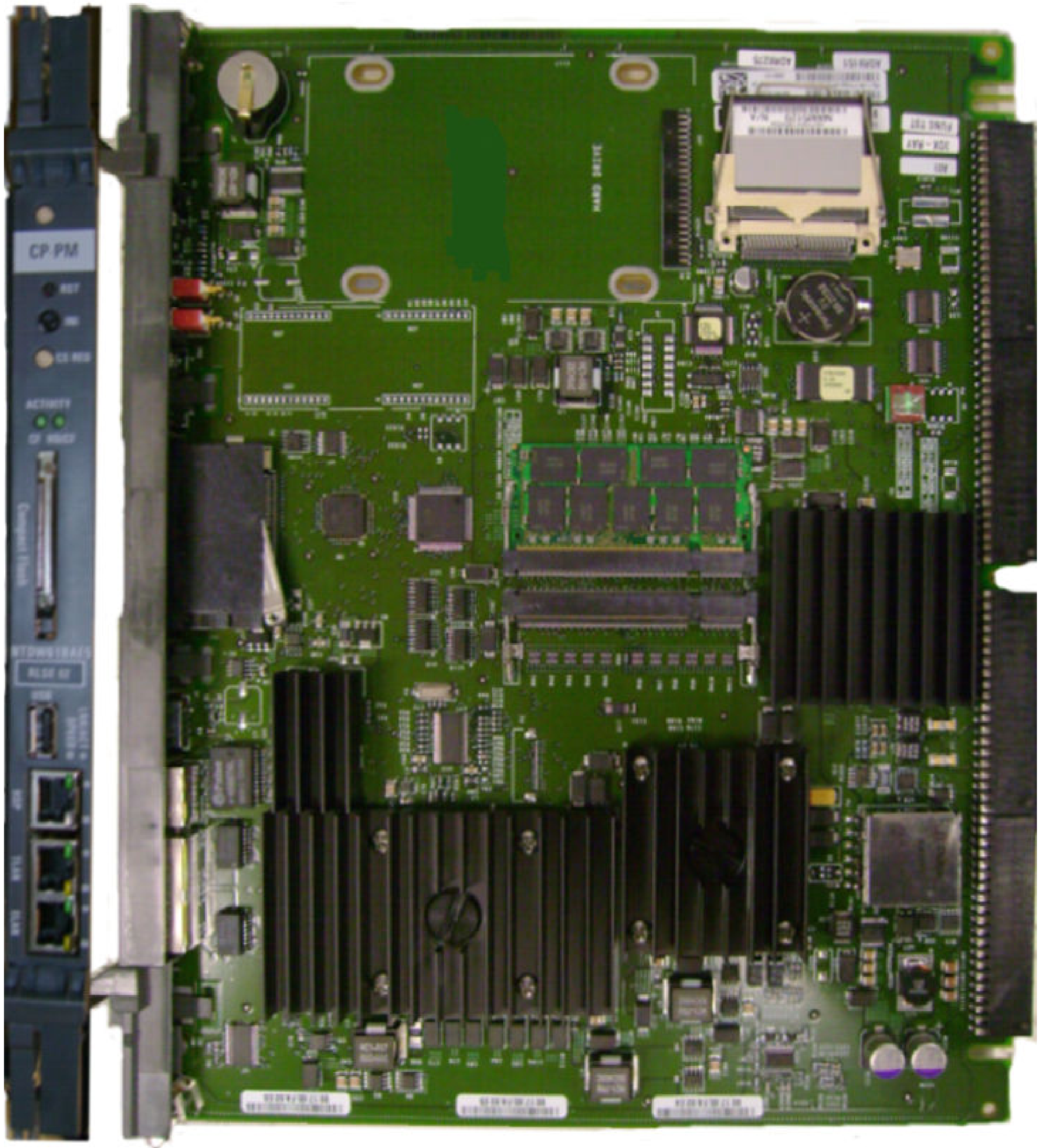

Figure 11: CP PM card NTDW61

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
- 1 GB 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

The 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.

Co-resident Call Server and Signaling Server hardware components

You can deploy a Co-resident Call Server and Signaling Server (Co-res CS and SS) with various hardware platforms. The Co-res CS and SS hardware platforms run the Linux Base Operating System. The following Servers support the Co-res CS and SS configuration:

- Common Processor Pentium Mobile (CP PM)
- Common Processor Dual Core (CP DC)
- Common Processor Media Gateway (CP MG)
- Commercial off-the-shelf (COTS) servers
    - IBM x3350 server (COTS2)
    - Dell R300 server (COTS2)
    - HP DL360 G7 server (Avaya Common Server)

For a description of each Server hardware platform, see *Avaya Circuit Card Reference, NN43001-311*.

During hardware installation, the Server circuit cards (CP PM, CP DC, CP MG) are inserted into a Media Gateway cabinet or chassis. COTS2 and Common Servers are inserted into standard 19" racks. Each Co-res CS and SS requires a Gateway Controller in a Media Gateway. The Gateway Controller can be a Media Gateway Controller (MGC) card, or a Common Processor Media Gateway (CP MG) card. The CP MG card functions as a Co-res CS and SS, and a Gateway Controller while occupying slot 0 in a Media Gateway.

For more information about Co-res CS and SS, see *Avaya Co-resident Call Server and Signaling Server fundamentals, NN43001-509*. For information about installing or configuring Co-res CS and SS applications, see *Avaya Linux Base and Applications Installation and Commissioning, NN43001-315*

Media Gateway

Media Gateways provide basic telephony media services, including tone detection, generation, and conference to CS 1000E telephones. The Media Gateway houses IPE circuit cards and connectors for access to the Main Distribution Frame. The Media Gateway with Media Gateway Controller (MGC) supports digital trunk and PRI access to the PSTN and to other PBX systems. The Media Gateway also supports Avaya 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.

Media Gateway hardware

The term Media Gateway refers to the following Media Gateway hardware types:

- MG 1000E chassis (NTDU14) and Media Gateway Expander chassis (NTDU15)
- Option 11C Mini chassis (NTDK91) and Option 11C Mini Expander chassis (NTDK92)
- Option 11C cabinet (NTAK11)
- MG 1010 chassis (NTC310)
- IPE module (NT8D37) with MG XPEC card (NTDW20)

For a physical description of the Media Gateway hardware, see *Avaya Communication Server 1000E Overview, NN43041-110*

The Branch Office uses the same Media Gateway hardware. For more information about Branch Office, see *Avaya Branch Office Installation and Commissioning, NN43001-314* .

Functional description

The Media Gateway provides the following functionality:

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

Circuit cards

The following circuit cards are supported in Media Gateways:

- One Gateway Controller card is required in each Media Gateway chassis or cabinet. See Media Gateway Controller (MGC) card on page 68 or Common Processor Media Gateway (CP MG) card on page 71 for Gateway Controller card details.
- An MG XPEC card in each IPE module. SeeMedia Gateway Extended Peripheral Equipment Controller (MG XPEC) card on page 72 for card details.
- Server cards (CP PM, CP DC, CP MG).
- Voice Gateway Media Cards. See Voice Gateway Media Card on page 81 for card details.
- Intelligent Peripheral Equipment (IPE) cards. See Operating parameters on page 67 for specific cards supported on Media Gateways.

For more information about circuit cards, see *Avaya Circuit Card Reference, NN43001-311*.

Security device

The security device on the Media Gateway is a generic security device that allows Media Gateways to register with the CS 1000E Call Servers.

Control for the activation of features assigned to the CS 1000E system, including Media Gateways, is provided by the security device on the System Utility card in CP PIV systems, and the security dongle on all other supported hardware platforms.

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

Operating parameters

The Media Gateway operates under the direct control of the Call Server. Up to 50 Media Gateways can be configured on the Call Server.

To allow IP Phones to access digital media services, you must configure Media Gateways with Digital Signal Processor (DSP) ports. The MGC with DSP daughterboards can provide up to

128 DSP ports, or 256 DSP ports when configured as a PRI Gateway. The CP MG card is available with 32 or 128 DSP ports. You can install Voice Gateway Media Cards into a Media Gateway to provide additional DSP ports beyond the DSP port limit of the Gateway Controller card.

The Media Gateways support 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 and Avaya 3900 Series Digital Deskphones, and external systems that use digital line emulation, such as Avaya 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
- Avaya 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 Media Gateways with MGC.

Media Gateway Controller (MGC) card

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

The 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 Call Server.

The MGC card supports up to 128 DSP ports with the two expansion sites to accommodate digital signal processor daughterboards. 128-port DSP daughterboard NTDW78, 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 amount of 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 *Avaya 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, or as backup TPS in case of failures.

Figure 12: Media Gateway Controller card (NTDW60) on page 70 shows the MGC faceplate and MGC circuit card (with two DSP daughterboards installed).

The MGC card (NTDW98) contains a metal faceplate for enhanced EMC containment. The NTDW98 MGC card is designed for use in a Media Gateway 1010 chassis.

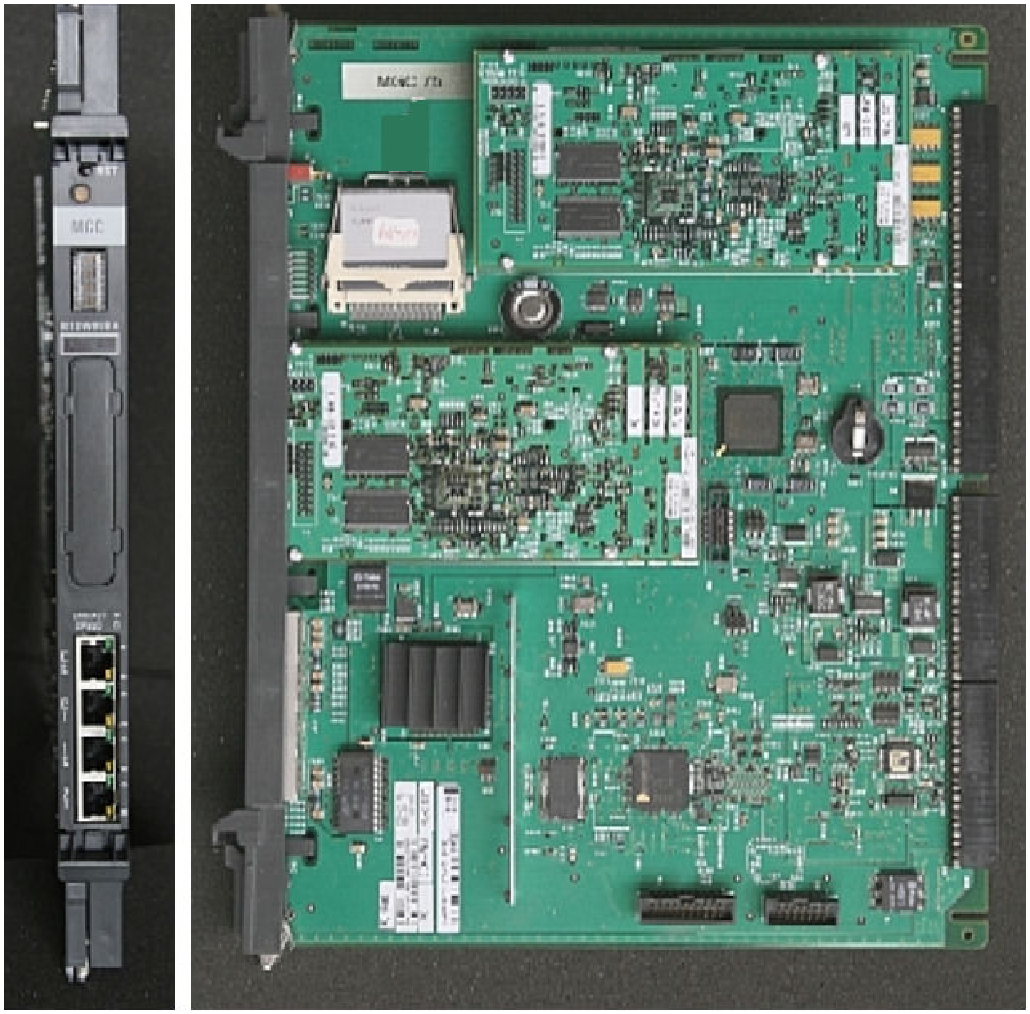

Figure 12: Media Gateway Controller card (NTDW60)

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

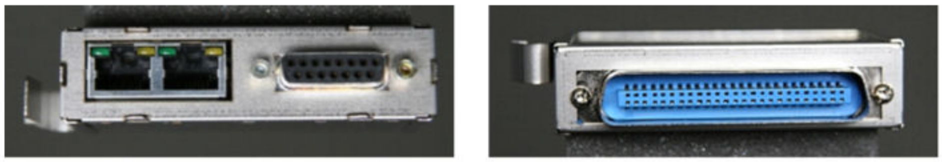

Figure 13: NTDW63 MGC adapter for Option 11C

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

Important:

The MGC is a gateway controller that replaces the SSC in a Media Gateway. It also reduces the need for separate Voice Gateway Media Cards with the use of onboard DSP daughterboards. The MGC-based Media Gateway supports PRI/PRI2/DTI/DTI2 trunks, BRI trunks, D-channels, and clock controllers. The MGC remote SDI feature reduces the need for separate Terminal Servers.

For more information about the MGC card, see *Avaya Circuit Card Reference, NN43001-311*.

Common Processor Media Gateway (CP MG) card

The hardware for the Common Processor Media Gateway (CP MG) card consists of integrating a Common Processor, a Gateway Controller, and non-removable Digital Signal Processor (DSP) resources into a single card for use in a Communication Server 1000E system. The CP MG card design is based on the CP PM card and MGC card with DSP daughterboards. The CP MG card is available in two versions:

- NTDW56BAE6 - CP MG card with 32 DSP ports
- NTDW59BAE6 - CP MG card with 128 DSP ports

The CP MG card provides improvements in port density and cost reductions by functioning as a Co-resident Call Server and Signaling Server (Co-res CS and SS) and a Gateway Controller with DSP ports while only occupying one slot in a Media Gateway cabinet or chassis. The CP MG card occupies the system controller slot 0 in a Media Gateway.

The CP MG card includes the following components and features:

- Intel EP80579 integrated processor, 1200 Mhz (Common Processor)
- 2 GB DDR2 RAM (expandable to 4 GB)
- 160 GB SATA hard drive
- USB to 1-wire bridge chip from Intel processor to security device
- One faceplate USB 2.0 port for software installations and upgrades
- Two faceplate Common Processor TTY serial ports
- Mindspeed Chagall-2 processor M82515 to run Avaya proprietary software (MGC, tone and conference)
- SPI flash for BIOS storage
- Compact Flash card (ATA) for Chagall-2 file system
- Onboard flash boot-ROM
- Three backplane TTY ports, two from Chagall-2, one from Intel Common Processor
- Mindspeed Picasso M82710 or Matisse M82910 for VoIP or DSP resources
- Ten port embedded fast Ethernet switch
- One faceplate 100 BaseT T-LAN port
- One faceplate 100 BaseT E-LAN port
- One backplane 100 BaseT T-LAN port
- One backplane 100 BaseT E-LAN port
- Faceplate status LED and card reset buttons
- In-rush power controller to support hot-plug

The Gateway Controller component of the CP MG card is based on the same architecture as the MGC card. For more information about the Gateway Controller architecture, see Media Gateway Controller (MGC) card on page 68. The Gateway Controller component of the CP MG card registers with the Call Server with an Internet Protocol Media Gateway (IPMG) type of MGS. The Communication Server 1000E system uses a common Gateway Controller loadware for the MGC, MG XPEC, and MGS.

For more information about the CP MG card, see *Avaya Circuit Card Reference, NN43001-311*.

Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card

You can convert NT8D37 Communication Server 1000M and Meridian 1 large system IPE modules into Communication Server 1000E Media Gateways with the Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card. The MG XPEC card provides a solution to migrate IPE modules from a Meridian 1 TDM system, or CS 1000M system to a CS 1000E system. The MG XPEC card converts one IPE module into two Media Gateway shelves (type MGX) for use in a CS 1000E system.

The NTDW20 MG XPEC card replaces the NT8D01 controller card in the controller slot of a NT8D37 IPE module. The MG XPEC card is a dual card assembly that contains a motherboard and a daughterboard. Each board of the dual assembly contains 192 non-removable Digital Signal Processor (DSP) ports. The MG XPEC card is essentially equivalent to two Media Gateway Controller (MGC) cards. The MG XPEC motherboard controls slots 0 to 7 on the left half of the IPE module, and the MG XPEC daughterboard controls slots 0 to 7 on the right half of the IPE module.

For information about converting an IPE module into Media Gateways with the MG XPEC card, see *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering, NN43021-220*.

Network connections

The ELAN of the Media Gateway can reside in a separate layer 3 subnet from that of the Call Server ELAN. When connecting the Media Gateway to the ELAN through a Layer 3 switch the connection from the Call Server to the Media Gateway must have a round trip delay of less than 80 msec and have a packet loss of less than 0.5 % (0% recommended).

Figure 14: Redundant network connections with MGC Dual Homing on page 74 is a schematic representation of redundant network connections for Media Gateways with MGC. The call servers, signaling servers, switches, and chassis can be any of the supported types.

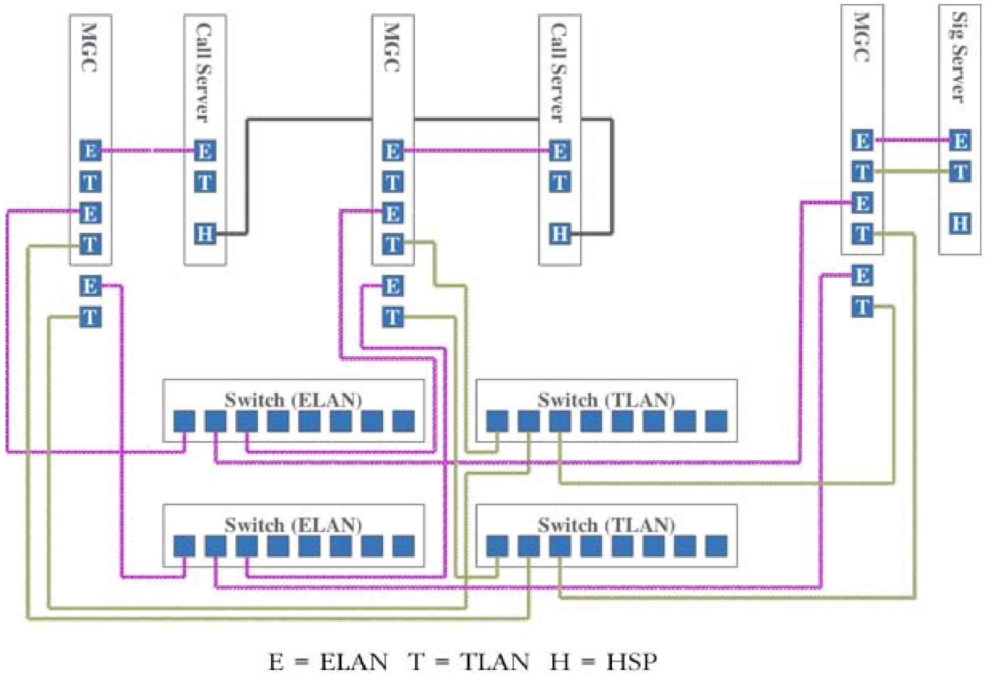

Figure 14: Redundant network connections with MGC Dual Homing

The separate LAN subnets that connect the Media Gateway 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 Media Gateways. 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. The HSP is not supported on Co-res CS and SS configurations.

Avaya CS 1000 Media Gateway 1000E

The CS 1000E Call Server can connect to and control a maximum of 50 Avaya CS 1000 Media Gateway 1000Es (Avaya MG 1000E) with optional Media Gateway Expanders. Each Avaya MG 1000E houses a Gateway Controller and contains four slots for IPE cards. Each MG 1000E supports an optional Media Gateway Expander through copper connections for a total of eight slots.

Physical description

The following sections describe the front and rear components of the MG 1000E (NTDU14).

Front components

Figure 15: Front components in the MG 1000E (NTDU14) on page 75 shows the Media Gateway with the front cover removed. Note the following:

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

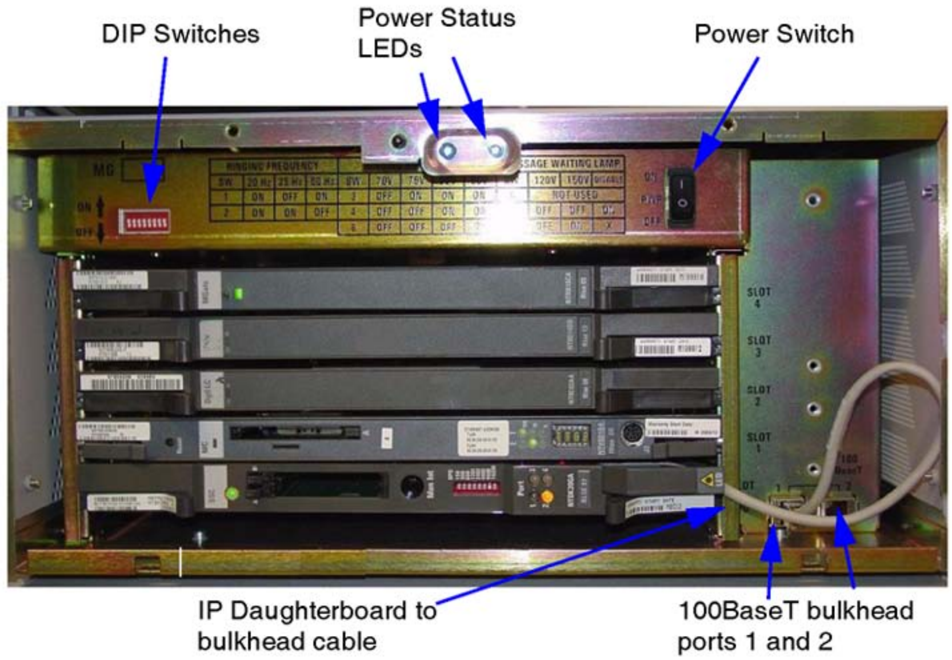

Figure 15: Front components in the MG 1000E (NTDU14)

Rear components

Figure 16: Rear components in the MG 1000E on page 76 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 card to other system components.
- The Attachment Unit Interface (AUI) is used with 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.

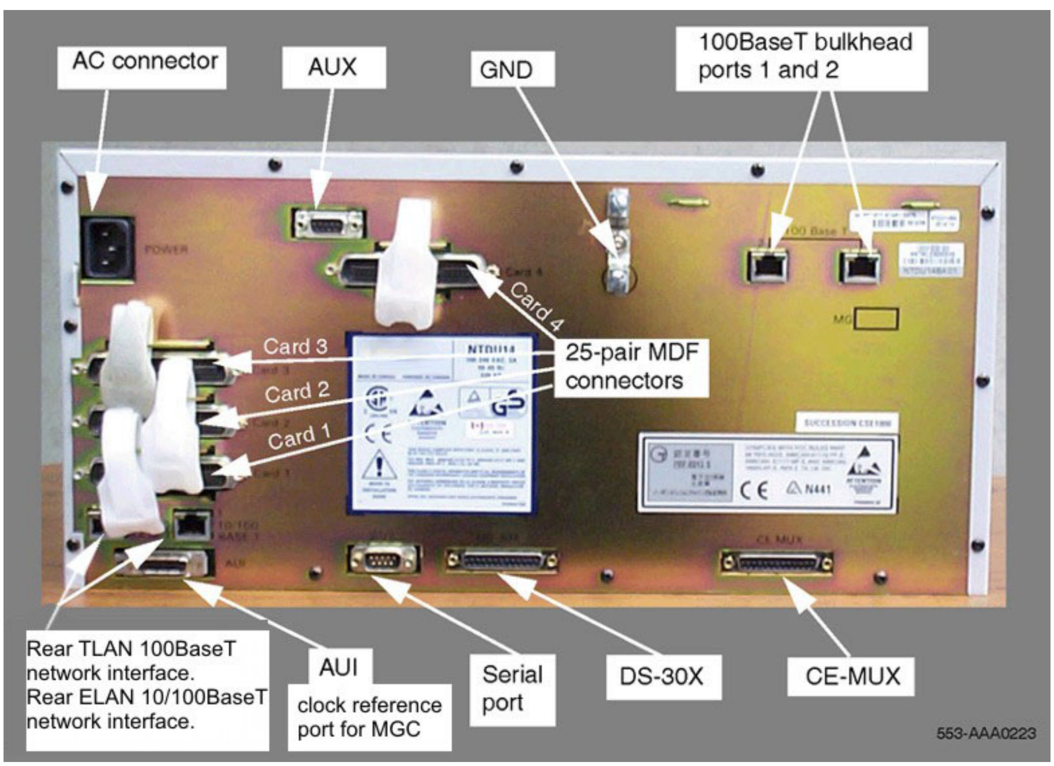

Figure 16: Rear components in the MG 1000E

Media Gateway Expander

The Media Gateway Expander supports up to four circuit cards. The Media Gateway Expander can directly support the Server cards. A Gateway Controller 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 17: Media Gateway Expander (NTDU15) on page 77 shows the Media Gateway Expander (NTDU15).

Figure 17: Media Gateway Expander (NTDU15)

Rear components

Figure 18: Rear components in the Media Gateway Expander on page 78 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.

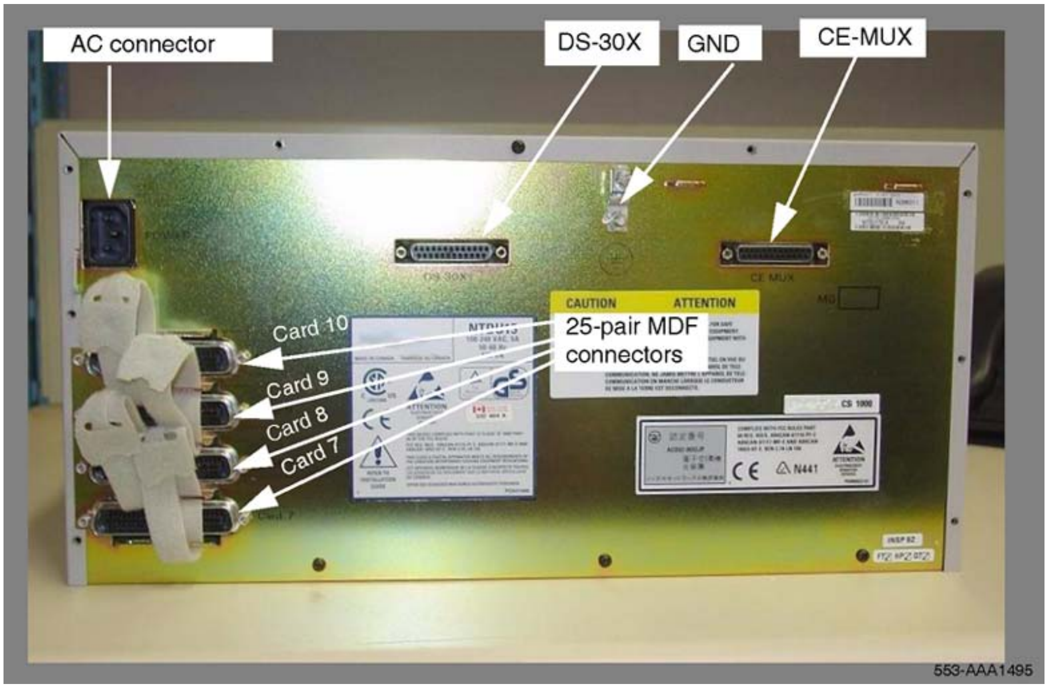

Figure 18: Rear components in the Media Gateway Expander

Media Gateway 1010

The Media Gateway 1010 (MG 1010) is a rack mount Media Gateway chassis that provides a larger amount of card slots than a MG 1000E with Media Gateway Expander. The CS 1000E Call Server can connect to and control a maximum of 50 MG 1010s. Each MG 1010 provides a dedicated Gateway Controller slot, two dedicated Server card slots, and ten slots for IPE cards. The MG 1010 is a single chassis that can provide more processing power and card capacity than a MG 1000E with the optional Media Gateway Expander.

Physical description

The Media Gateway 1010 chassis (NTC310AAE6) consists of:

- MG 1010 rack mount kit (NTC316AAE6)
- backplane assembly (NTC31002)
- Media Gateway Utility (MGU) card (NTC314AAE6)
- power supply, maximum of two with load sharing (NTC312AAE6)
- blower fans, N+1 arrangement for redundant cooling (NTC320AAE6)
- air filter (NTC315AAE6)
- front cover with EMC containment and a window to view status LEDs
- MG 1010 serial cable kit (NTC325AAE6)

Metal faceplate Server cards and Gateway Controller cards are required for enhanced EMI containment. Avaya recommends you to use metal CP PM card (NTDW99) and metal MGC card (NTDW98) in a MG 1010. All CP MG and CP DC cards contain metal faceplates.

The following sections describe the front and rear components of the MG 1010 (NTC310).

Front components

Figure 19: Front components in the MG 1010 on page 79 shows the Media Gateway 1010 without the front cover. Note the following:

- Ten IPE card slots
- Two Server card slots
- One Gateway Controller card slot
- One Media Gateway Utility (MGU) card provides LED status, ringing, message waiting voltage, dual homing Ethernet cable ports, and serial cable ports
- One metal divider in chassis to separate MGU, Server cards, and Gateway Controller card from the IPE cards.

Figure 19: Front components in the MG 1010

Figure 20: MG 1010 front cover on page 80 shows the MG 1010 with the front cover. Note the following:

- Window to view LED status of all cards
- Decorative cover provides additional EMC shielding
- Two locking latches in top corners of front cover.

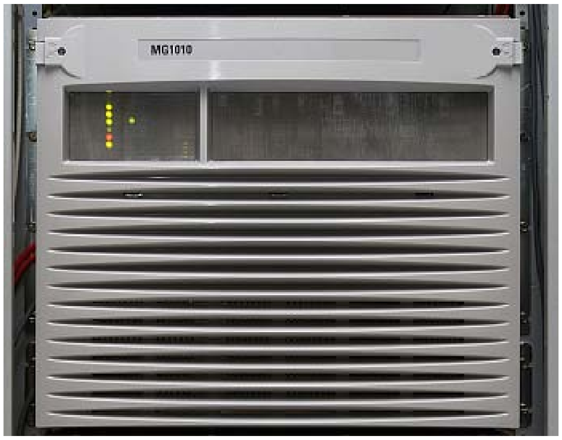

Figure 20: MG 1010 front cover

Rear components

Figure 21: Rear components of the MG 1010 on page 81 shows the rear components of the MG 1010. Note the following:

- Hot swappable redundant power supplies
- Hot swappable fans in a redundant N + 1 configuration for chassis cooling
- One DECT connector
- One AUX connector
- Ten MDF connectors

Figure 21: Rear components of the MG 1010

Voice Gateway Media Card

The Gateway Controller cards can provide a limited number of DSP ports. Voice Gateway Media Cards provide additional DSP ports for Media Gateways to translate between IP and TDM. Voice Gateway Media Cards provide interfaces that connect to the Telephony Local Area Network (TLAN) and Embedded Local Area Network (ELAN) subnets. The Voice Gateway Media Card is also known as a Media Card (MC).

Figure 22: Media Card on page 82 shows faceplate connectors and indicators on the 32-port line Media Card.

Figure 23: Media Card 32S on page 83 (MC32S) shows the faceplate connectors and circuit card of the 32-port secure Media Card.

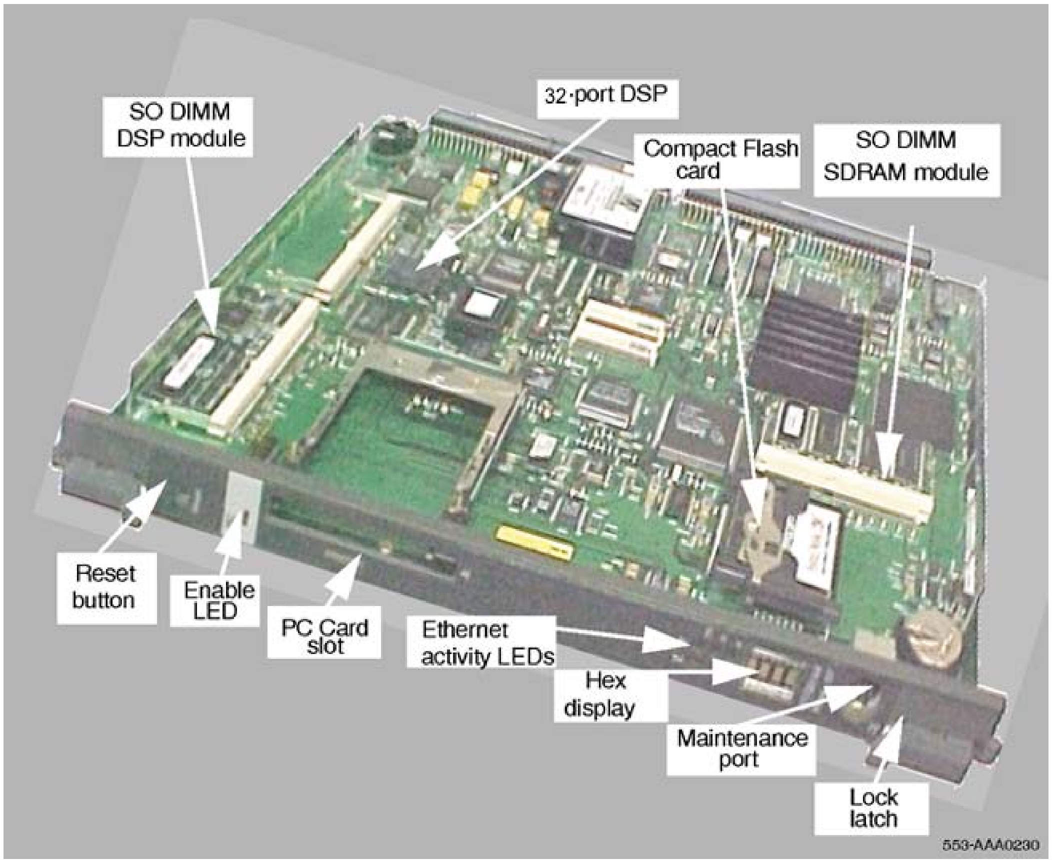

Figure 22: Media Card

Figure 23: 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 card.

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 *Avaya Security Management, NN43001-604*.

For more information about Media Card features or the IP Line application, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

Signaling Server

Main role

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

The hardware platforms supported as stand-alone Signaling Servers are CP PM, CP DC, and Commercial off-the-shelf (COTS) Servers. Available COTS servers are IBM x306m, IBM x3350, HP DL320 G4, HP DL360 G7, and Dell R300 servers.

The Communication Server 1000E Linux Platform Base includes many operational, performance, and security hardening updates. The User Access Control (UAC) introduces eight Linux groups to define user privileges. Central Authentication provides user authentication across the security domain with single password. The Emergency Account allows you to log on through the Command Line Interface (CLI) if both the Primary and Secondary UCM are offline. Secure File Transfer Protocol (SFTP) is the default file transfer protocol. You must explicitly identify FTP users, all users can use SFTP.

The Linux Platform Base operating system installs on the Signaling Server and can run multiple applications, including:

- SIP and H.323 Signaling Gateways
- Terminal Proxy Server (TPS)
- Network Routing Service (NRS)
- SIP Line Gateway (SLG)
- Element Manager
- Application Server for Personal Directory (PD), Callers List (CL), Redial List (RL), and Unicode Name Directory (UND) for UNIStim IP Phones

SIP Line Gateway includes:

- SIP Line (SIPL)
- SIP Management Service, an Element Manager (EM) system management interface you use to configure and manage the SIP Line Service.

NRS includes:

- H.323 Gatekeeper
- SIP Proxy Server

- SIP Redirect Server
- NCS

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

CP PM Signaling Server physical description

Figure 24: CP PM Signaling Server card and faceplate on page 86 shows the CP PM Signaling Server. Two versions of the CP PM Signaling Server are available, each designed for use in the CS 1000M or CS 1000E chassis, however they are functionally identical. NTDW61 single slot CP PM card for Media Gateway chassis, or Option 11C cabinets. NTDW66 double wide faceplate CP PM card for CS 1000M Large System IPE cubes.

The NTDW99 CP PM card is an update to the NTDW61 CP PM card. It is functionally identical, however it contains a metal faceplate for enhanced EMC containment. The NTDW99 CP PM card is designed for use in a Media Gateway 1010 chassis.

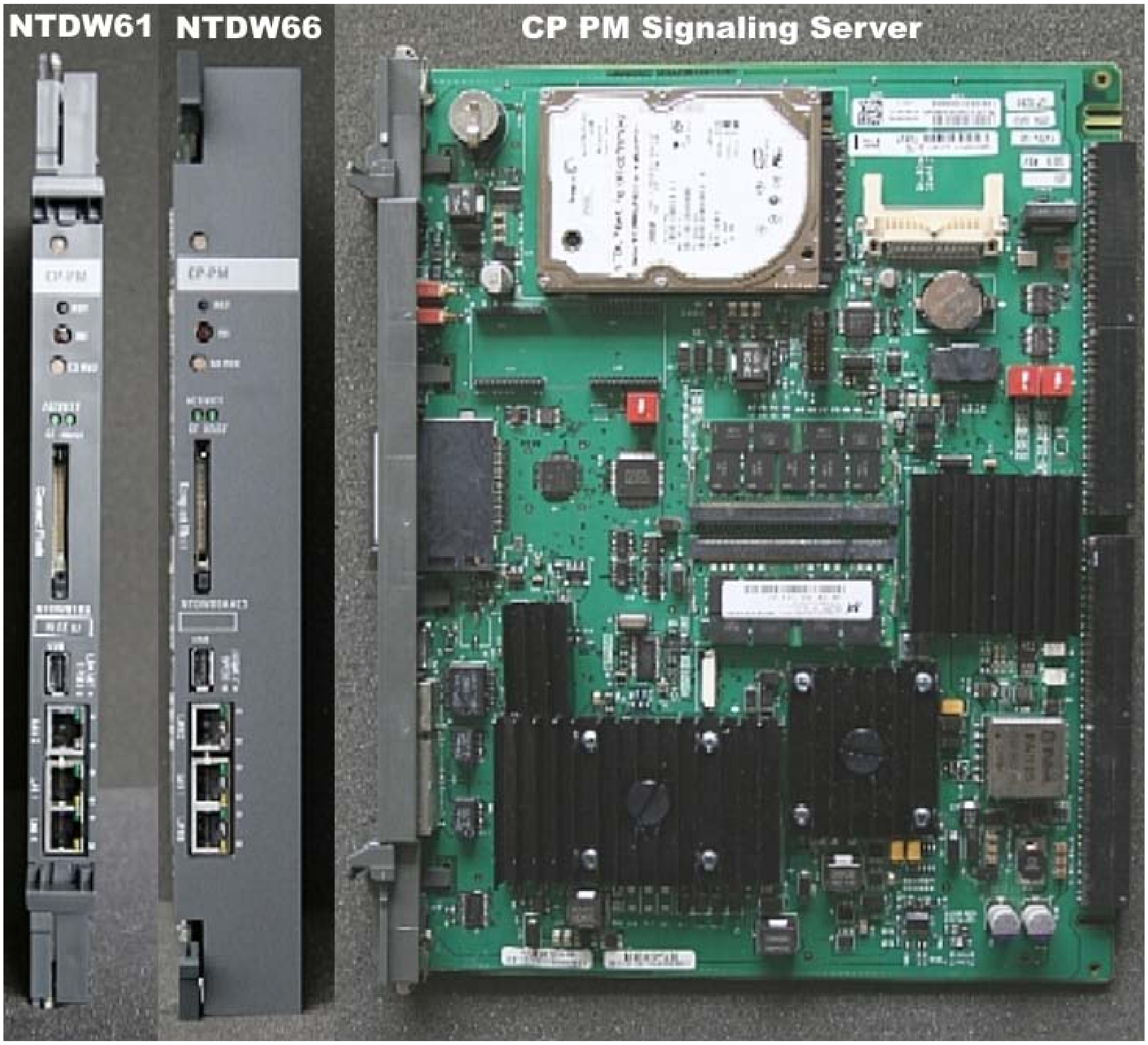

Figure 24: CP PM Signaling Server card and faceplate

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 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 2 GB of SDRAM. CP PM card on page 63 provides further details the CP PM hardware components.

CP DC Signaling Server physical description

The Common Processor Dual Core (CP DC) card is a Server card for use in Communication Server 1000E and Communication Server 1000M systems. The CP DC card is designed to replace the Common Processor Pentium Mobile (CP PM) card. The CP DC card contains a dual core AMD processor and upgraded components which can provide improvements in processing power and speed over the CP PM card. The CP DC card requires the Linux Base Operating System, and supports Co-resident Call Server and Signaling Server, or stand-alone Signaling Server configurations.

The CP DC card is available in two versions:

- NTDW53AAE6 - single slot metal faceplate CP DC card (CS 1000E).
- NTDW54AAE6 - double slot metal faceplate CP DC card (CS 1000M).

The CP DC card provides performance improvements in MIPS, maximum memory capacity, and network transfer rate, and occupies one IPE slot in a Media Gateway.

CP DC Signaling Server hardware components

The CP DC card includes the following components and features:

- AMD Athlon 64 X2 1.8 Ghz dual core processor
- 2 GB DDR2 RAM
- 160 GB SATA hard drive
- Three faceplate USB 2.0 ports for software installations, upgrades, patching, and USB keyboard and mouse support
- One faceplate VGA port for monitor support
- Two faceplate Gigabit Ethernet ports
- Faceplate status LED and card reset buttons

You can use a USB 2.0 storage device to install or upgrade the Linux Base Operating System. The CP DC card does not support Compact Flash (CF) cards.

An NTDW5309E6 2 GB memory upgrade kit is available for Media Application Server (MAS) deployments. This upgrades a CP DC card to a total of 4 GB DDR2 RAM.

COTS Signaling Server physical description

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

COTS Signaling Server hardware components

IBM x306m and HP DL320 G4 (COTS1) 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

IBM x3350 and Dell R300 (COTS2) and HP DL360 G7 (Common Server) Signaling Servers provide the following features:

- Intel Xeon quad-core processor (minimum of 2.4 GHz)
- Two simple swap Serial-attached SCSI (SAS) or SATA hard drives (minimum of one 75 GB hard drive)
- 4 GB of RAM PC2-5300 DDR II with error correcting code (ECC)
- Two 10/100/1000BaseT Ethernet ports
- Four USB 2.0 ports
- 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)
- SIP Line Gateway (SLG)
    - SIP Line
    - SIP Management Service
- H.323 Gateway (Virtual Trunk)

- H.323 Gatekeeper
- Network Routing Service (NRS)
    - SIP Redirect Server
    - SIP Proxy Server
    - SIP Registrar
    - NRS Manager
- CS 1000 Element Manager
- Application Server for the Personal Directory, Callers List, Redial List, and Unicode Name Directory 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 44: Elements in Signaling Server on page 231. For detailed Signaling Server engineering rules and guidelines, see Signaling Server algorithm on page 271. For more information about H.323, SIP Trunking NRS and SIP Proxies see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313* and *Avaya Network Routing Service Fundamentals, NN43001-130*.

For more information about SIP Line and IP Line, see *Avaya SIP Line Fundamentals, NN43001-508* and *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

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 Media Gateways
- supports key software components (see Software applications on page 88)

Operating parameters

The Signaling Server provides signaling interfaces to the IP network using software components that run on the Linux Base operating system. For more information about the Signaling Server Linux Base, see *Avaya Linux Platform Base and Applications Installation and Commissioning, NN43001-315*.

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 calculations on page 284).

Terminal Server

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

Important:

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

Physical description

Figure 25: Terminal Server on page 90 shows the Terminal Server.

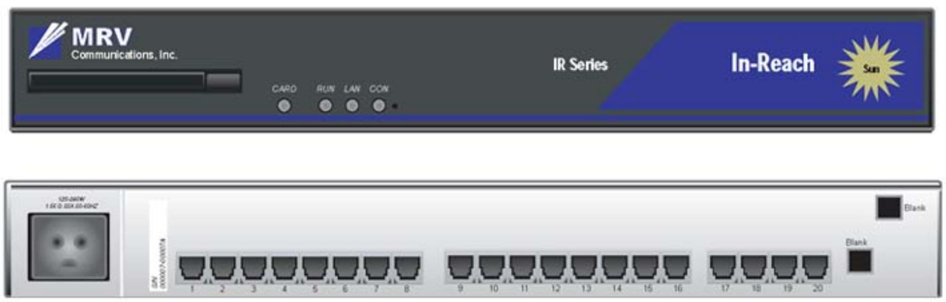

Figure 25: Terminal Server

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

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

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.

Important:

Currently, the CS 1000E only supports the MRV IR-8020M commercial Terminal Server.

Layer 2 switch

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 26: ELAN Layer 2 switch (Ethernet Routing Switch 2550T) on page 92 shows an example of an ELAN Layer 2 switch.

Figure 26: ELAN Layer 2 switch (Ethernet Routing Switch 2550T)

TLAN Layer 2 switch

To provide Layer 2 connections on the TLAN subnet, Avaya recommends the Ethernet Routing Switch 2526T, 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 27: TLAN Layer 2 switch (Ethernet Routing Switch 2526T) on page 92 shows the Ethernet Routing Switch 2526T.

Figure 27: TLAN Layer 2 switch (Ethernet Routing Switch 2526T)

Operating parameters

These components must be supplied by the customer. For more information, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

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.

For more information, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Telephones

The CS 1000E system supports the following:

- IP Phones
    - IP Phone 2001
    - IP Phone 2002
    - IP Phone 2004
    - Avaya 2007 IP Deskphone
    - Avaya 1120E IP Deskphone
    - Avaya 1140E IP Deskphone
    - Avaya 1150E IP Deskphone
    - Avaya 1210 IP Deskphone
    - Avaya 1220 IP Deskphone
    - Avaya 1230 IP Deskphone
    - Avaya 2050 Mobile Voice Client (MVC)
    - Avaya 2050 IP Softphone
    - Avaya 2033 IP Conference Phone
    - WLAN Handset 2210, 2211 and 2212
    - IP Phone Key Expansion Module (KEM)
- analog (500/2500-type) telephones
- digital deskphones

- attendant consoles
- DECT handsets
- 802.11 Wireless LAN terminals

Component dimensions

All rack mount components fit in 19-inch racks. Table 4: Height dimension of CS 1000E components on page 94 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 4: Height dimension of CS 1000E components

Component	Height
NTDU62 CP PIV Call Server	3 U
Signaling Server (COTS)	1 U
NTDU14 Media Gateway	< 5 U
NTDU15 Media Gateway Expander	< 5 U
NTC310 Media Gateway 1010	9 U
MRV Terminal Server	1 U
Ethernet Routing Switch 2526T	1 U
Ethernet Routing Switch 2550T	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.).

Chapter 7: Configuration options

Contents

This chapter contains the following topics:

Introduction

The IP-distributed architecture of the Avaya Communication Server 1000E (Avaya CS 1000E) enables flexibility when it comes to component location. Given this flexibility, the Avaya 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 Media Gateways
    - 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.

Fax/Modem pass through

The Fax/Modem pass through feature provides a modem pass through allowed (MPTA) class of service (CLS) for an analog phone TN. MPTA CLS dedicates an analog phone TN to a modem or a Fax machine terminal. A connection that initiates from the dedicated TN, and/or calls that terminate at the dedicated TN through a Digital Signal Processor (DSP), use a G711 NO VAD codec on the Call Server.

Modem Pass through is a specific configuration of a G.711 VoIP channel that improves modem performance compared to standard VoIP configuration. Automatic switch to Voice Band Data (VBD) is a feature of the DSP; the DSP monitors the data stream to distinguish between voice and data calls. The DSP reconfigures to modem pass through mode when it determines the call is a modem call.

For modem calls between CS 1000 systems connected by analog and digital trunks, you must configure MPTA CLS on the Call Server of each CS 1000 system for analog units connected to modems. MPTA CLS configuration is necessary because the call setup negotiation is not done end to end as it is for virtual trunks. If the analog unit on one Call Server uses MPTA CLS and the analog unit on the other Call Server uses modem pass through denied (MPTD) CLS, the modem call fails.

MPT CLS is supported by the G.711 codec only; MPT CLS includes no other codecs. The packet interval for G.711 codec is set to 20 msecs in MPT.

The maximum speed supported for modem and fax is 33.6 Kbps. This limit is imposed by the analog line card.

MPT allows CS 1000 to support the following:

- modem pass through
- Super G3 (SG3) fax at V.34 (33.6 Kbps)
- V.34 rate (33.6 Kbps) modems
- Fax machines that support V.17, V.27, V.29, and V.34 protocols

For interface commands, responses, and definitions for MPT see Table 5: Interface commands and responses on page 96.

Table 5: Interface commands and responses

Command prompt	User response	Description
CLS	MPTA	Turn on the MPT feature.
CLS	MPTD	Turn off the MPT feature.

Note:

CLS MPTA and MPTD is included in LD10 for analog line card units.

For information about feature packaging requirements see Table 6: Feature packaging requirements on page 97.

Table 6: Feature packaging requirements

Package mnemonic	Package number	Package description	Package type (new, existing, or dependency)	Applic able market
Softswitch	402	Identifies a softswitch system.	Existing	All
Media Gateway	403	Identifies a system that is equipped with Media Gateways.	Existing	All

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
- V.34 rate (33.6 Kbps)

Performance degrades significantly with packet loss.

Modem and fax traffic performance is improved with Modem/Fax Pass Through Allowed (MPTA) class of service in an analog phone TN. The call server selects a G.711 codec with no Voice Activity Detection (VAD) to setup the call. Call Server disconnection or slow modem and fax data rates caused by closing the voice connection and reconnecting with T.38 codec have been eliminated with MPTA.

The CS 1000E supports Modem Pass Through and Super G3 (SG3) fax at V.34 (33.6 Kbps). You can configure MPTA for a TN, and the codec used by the DSP for that TN will be G.711 NO VAD only.

Performance degrades significantly with packet loss. Packet loss and latency in modem pass through mode can degrade connectivity rate, throughput rate and drop calls.

If you plan to route modem calls through analog trunks, you must configure the line cards for the modems and the analog trunks to route in the same Media Gateway. Delay issues can also be addressed by using a baud rate of 2400 baud or lower on the modem, provided the customer needs can be met at the low baud rate.

Avaya recommends you use digital trunks and only hardware modems in large systems which require modems and a large number of trunks. Do not deploy software modems across different Media Gateways regardless of the trunk type. Replace software modems with hardware modems or other IP interfaces.

The following hardware modems have been tested with the Communication Server 1000E using digital trunks:

- U.S. Robotics 5637
- U.S. Robotics 5685E
- U.S. Robotics 5699B

The guidelines listed above must be considered for upgrades to a CS 1000E system from a Option 11C or CS 1000M. Some configuration changes may be necessary.

Important:

Avaya has conducted extensive but not exhaustive tests of modem-to-modem calls, data transfers, and file transfers between a CS 1000E and Media Gateway, using Virtual Trunks and PRI tandem trunks. While all tests have been successful, Avaya 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 Avaya representative for more information.

Recommendations for Fax configuration in the CS 1000 system

Fax settings and performance over Voice over IP (VoIP) solutions vary depending on the network configuration. In order to achieve a successful faxing environment, the VoIP solution must be engineered properly. This section describes the configuration and network design aspects that need to be taken into consideration when implementing faxing in VoIP solutions.

CODECs:

T.38:

- Older fax machines use V.21
- For lower speeds such as V.21, T.38 protocol should be used in the VoIP segments of the call

Modem pass-through (G.711):

- Newer fax machines use modem protocols to achieve higher speeds (V.34)
- The Modem pass-through feature is intended for modems and high speed faxes employing V.34, using clear channel G.711 over the VoIP segments of the call.
- The Modem pass-through feature detects the phase reversal tone negotiation used for higher speeds and instructs the Digital Signal Processors (DSPs) involved in the call to disable echo cancellation and all other non-linear components.
- The Modem Pass-Through Denied (MPTD) class of service (CoS) for analog fax lines allows the DSPs to switch between T.38 protocol for lower speeds and G.711 protocol for higher speeds.
- The MPTD CoS must be used on analog line card (ALC) units connected to fax machines when there are trunk cards present in the IP Media Gateway (IPMG) and other IPMGs that connect to the TN. The MPTD class of service is required in order to support T.38 and V.34 faxes that could originate or terminate from/to the TN.
- In order for MPTD CoS to work, a system bandwidth strategy of BQ (Best Quality) must be used.
- The Modem Pass-Through Allowed (MPTA) CoS should be used only for modems, as it will force all calls to use G.711 protocol.
- When going to the TN, there is no control over the far-end; however, if the far-end supports T.38 and Modem pass-through, speeds of 33.6 Kbps should be achievable.

Important:

Fax performance at higher speeds (33.6 Kbps) requires that all network elements are properly engineered to support it. When high speed faxes cannot be achieved with a consistent success rate, Avaya recommends that the fax units be set at a lower speed (14.4 Kbps).

Typical scenarios

Typical scenarios for faxing in a CS 1000E solution are:

- two faxes connected to analog lines in the same Media Gateway Controller (MGC)
- two faxes connected to analog lines in different MGCs of the same CS 1000E system
- one fax connected to an analog line in an MGC, to an IP trunk, to a remote system with a fax
- one fax connected to an analog line in an MGC, to an analog trunk in the same MGC, and then to a TN (Local or Long Distance (LD)) fax

- one fax connected to an analog line in an MGC, to a digital trunk in the same MGC, and then to a TN (Local or LD) fax
- one fax connected to an analog line in an MGC, to a digital trunk in a different MGC of the same CS 1000E system, to a TN (Local or LD) fax

Note:

The following scenario is not supported for faxing: one fax connected to an analog line in an MGC, to an analog trunk in a different MGC of the same CS 1000E system, to a TN (Local or LD) fax.

Recommendations for faxing in a data network

Depending on the fax scenario used, fax calls can traverse the IP network, either internally (e.g., MGC to MGC) or externally (e.g., IP trunks). It is important to engineer the data network to support the following:

- Media card configuration:
    - G.711/T.38 codecs
    - 20 ms packet size
- Round trip delay must be less than 50 ms
- Packet loss must be less than 0.5%
- V.34 rate (33.6 Kbps) can be used as long as the far end supports the Modem pass-through feature
- Mean jitter is less than 5 ms

Important:

Performance degrades significantly with packet loss (must be less than 0.5%) and when the delay (round trip) is greater than 50 ms and the mean jitter is greater than 5 ms.

Important:

Avaya conducted extensive but not exhaustive test of fax calls in different scenarios. While all tests succeeded, Avaya cannot guarantee that all fax brands can operate properly over all G.711 Voice over IP (VoIP) networks. Before you deploy faxes, test the fax within the network to verify reliable operation. Contact your system supplier or your Avaya representative for more information.

Configuration for analog line cards connected to faxes

The typical configuration recommended for analog line cards connected to faxes is as follows:

- MGCs:
  - Enable Modem pass-through mode in the Element Manager
  - Enable V.21 FAX tone detection in the Element Manager
  - Set Voice Gateway (VGW) trunks to the zone with Best Quality (BQ)
- Analog lines:
  - Use Class of Service FAXA to set the proper trunk capability for fax calls
  - Use Class of Service Modem Pass-Through Denied (MPTD). This setting allows lower speed faxes (up to 14.4 Kbps) to use T.38, and higher speed faxes to use G.711 clear channel (no echo cancellation, no non-linear DSP features)

Option 1: Campus-distributed Media Gateways

With multiple buildings in a campus, you can distribute Media Gateways across a campus IP network. Figure 28: Campus-distributed Media Gateways on page 102 shows Media Gateways distributed across multiple buildings in a campus setting.

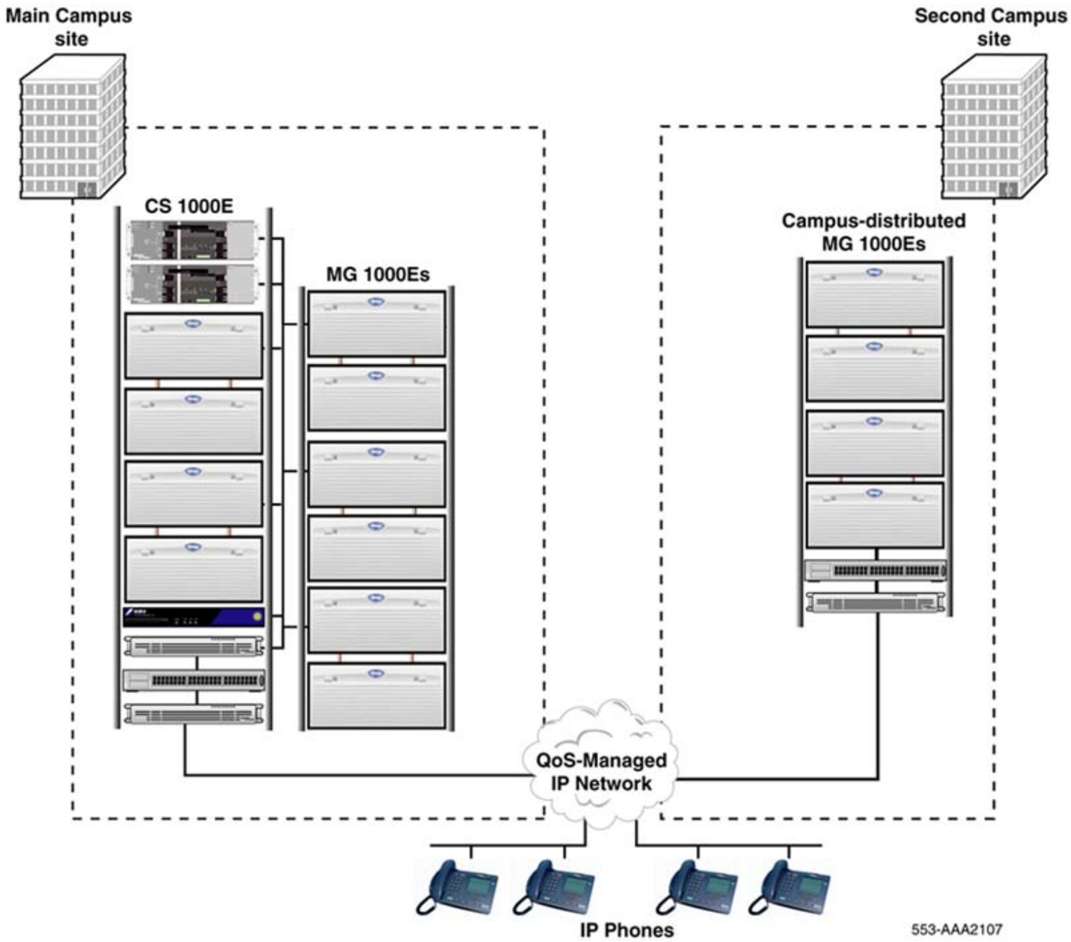

Figure 28: Campus-distributed Media Gateways

In this configuration, a CS 1000E system is installed at the main site, and additional Media Gateways 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 Media Gateway, see Media Gateway on page 66.

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 *Avaya 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 29: Campus Redundancy configuration on page 103 shows a CS 1000E system in a Campus Redundancy configuration. For more information, see *Avaya System Redundancy Fundamentals, NN43001-507*.

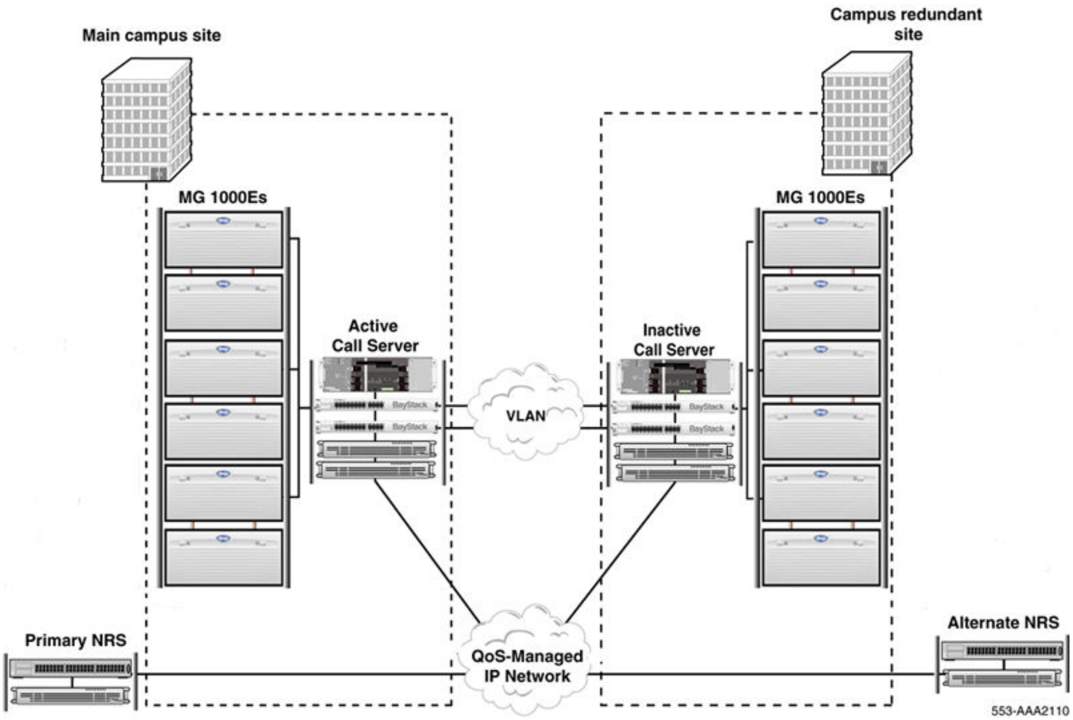

Figure 29: Campus Redundancy configuration

Option 3: Branch Office

The CS 1000E system supports the Branch Office feature, which provides central administration of Media Gateways at remote sites. Figure 30: Branch Office configuration on page 104 shows a CS 1000E system with an Media Gateway installed at a remote branch office.

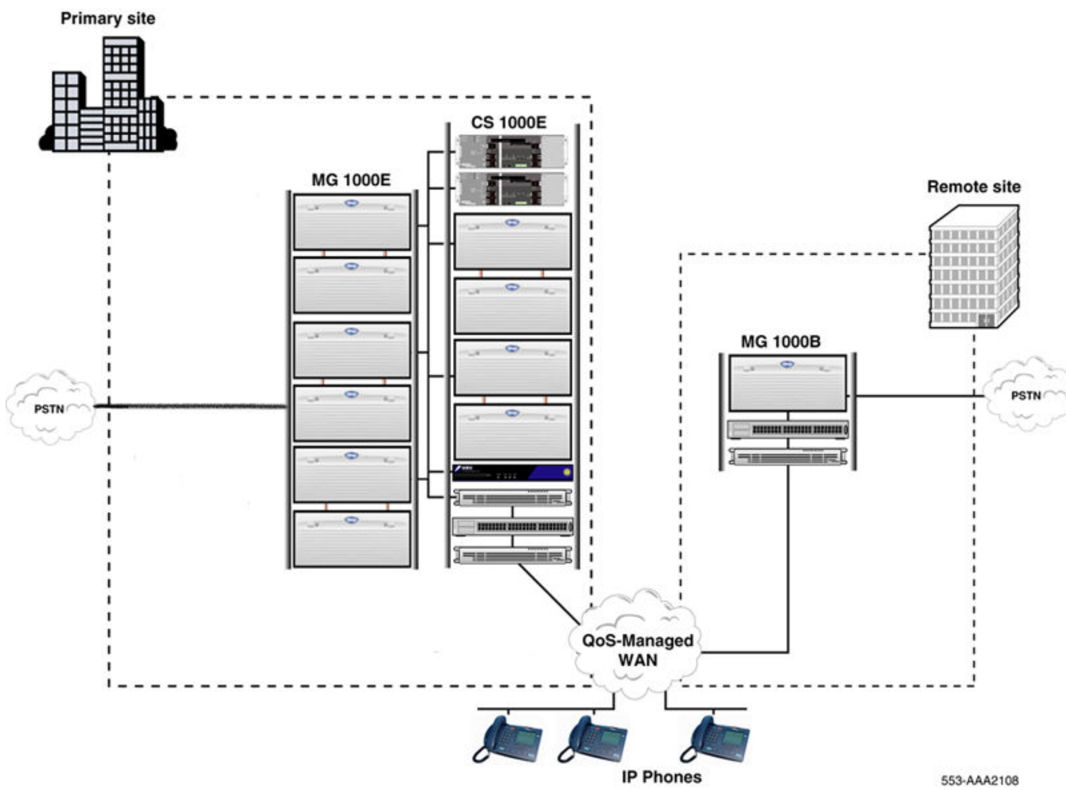

Figure 30: Branch Office configuration

In this configuration, the Branch Office Media Gateway is survivable. This ensures that telephone service remains available if the main office fails. For more information about Branch Office, see *Avaya Branch Office Installation and Commissioning, NN43001-314*. For more information about Survivable Media Gateway, see *Avaya System Redundancy Fundamentals, NN43001-507*.

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 31: Geographic Redundancy Survivable Media Gateway configuration on page 105shows the different paths of registration and replication.

TDM resources redundancy is achieved by Media Gateway and Voice Gateway Media Card Triple Registration on page 105.

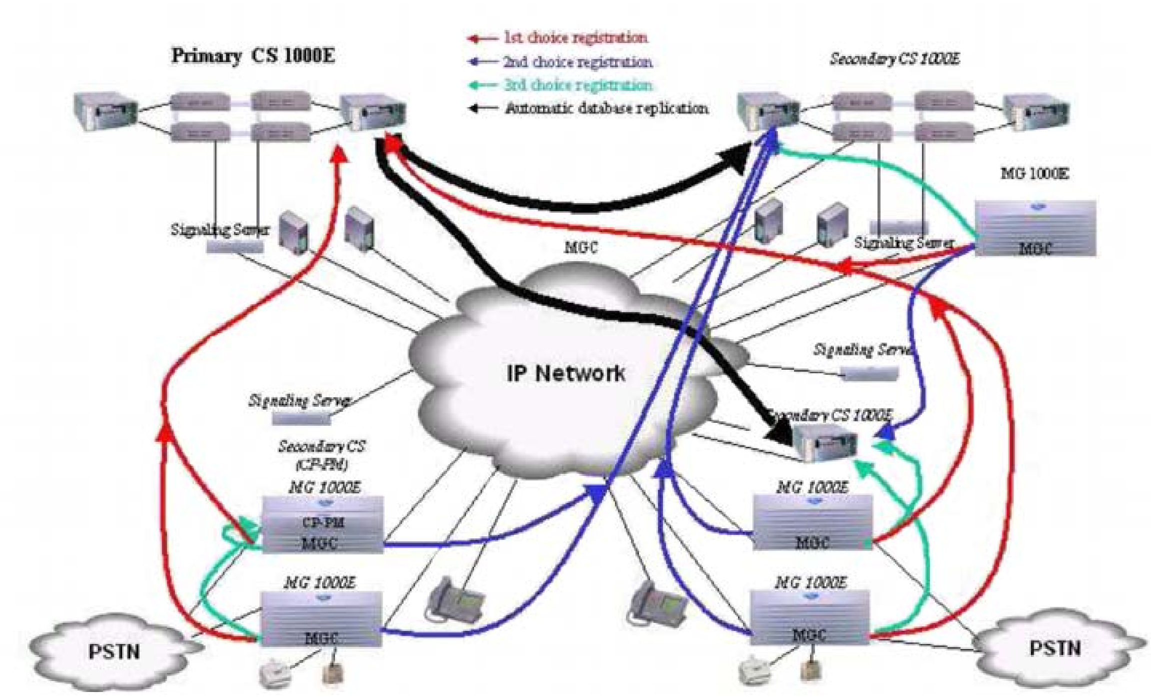

Figure 31: Geographic Redundancy Survivable Media Gateway configuration

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

Media Gateway and Voice Gateway Media Card Triple Registration

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

In normal mode, all Media Gateways 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 Media Gateway 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 Media Gateway reboots and registers to its configured alternate Call Server 1.

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

When the Media Gateway is registered to any of the alternate Call Servers, it continues to poll configured Call Servers. When the primary Call Server is detected, the Media Gateway 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 Media Gateway remains registered to an alternate Call Server until a command is entered.

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 Media Gateway and Voice Gateway Media Card registering on different alternate Call Servers during a primary Call Server failure, the Media Gateway sends a message to Voice Gateway Media Card in each Media Gateway to register with the same alternate Call Server that the Media Gateway is registered with.

Chapter 8: Planning reliability strategies

Contents

This chapter contains the following topics:

Introduction

Reliability in the Avaya Communication Server 1000E (Avaya 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 Avaya 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
- 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 on page 108. See Call Server failure on page 108.
- Network failure on page 109. See Network failure on page 109.
- Signaling Server failure on page 110. See Signaling Server failure on page 110.
- NRS failure on page 111. See NRS failure on page 111.
- NRS failure fail-safe on page 112. See NRS failure fail-safe on page 112.

Call Server failure

Figure 32: Call Server failure on page 109 shows a network-wide view of Call Server failure.

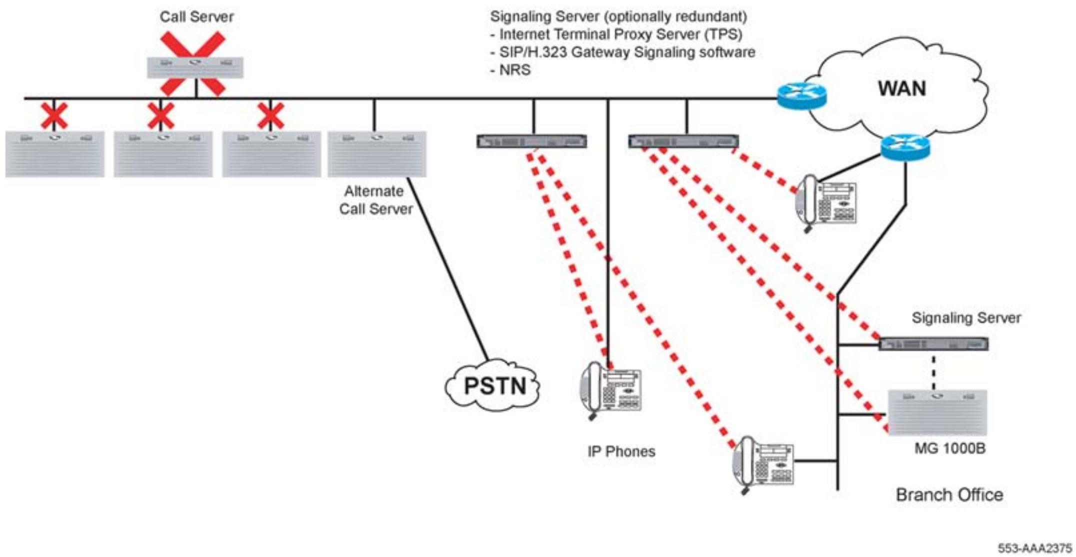

Figure 32: Call Server failure

Alternate Call Server

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

When planning reliability strategies, provision one Media Gateway 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 32: Call Server failure on page 109, the Media Gateway 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 Media Gateway cards, such as analog and PRI cards.

Network failure

Figure 33: Network failure with Survivable Media Gateway on page 110 illustrates a network failure with Survivable Media Gateway.

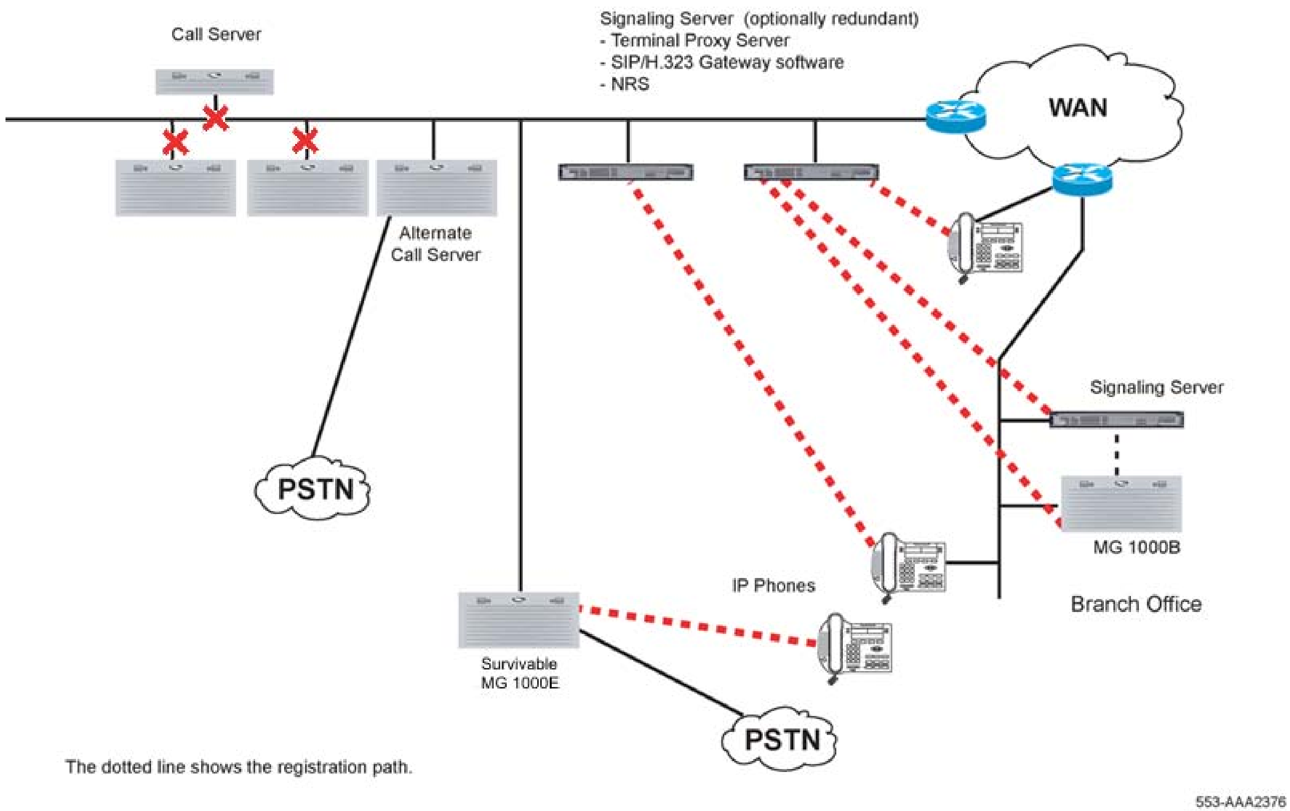

Figure 33: Network failure with Survivable Media Gateway

Campus-distributed Media Gateway in Survival Mode

Media Gateways 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 Media Gateway, consider the location of critical telephones and trunks.

Signaling Server failure

Figure 34: Signaling Server failure on page 111 illustrates the failure of a Signaling Server.

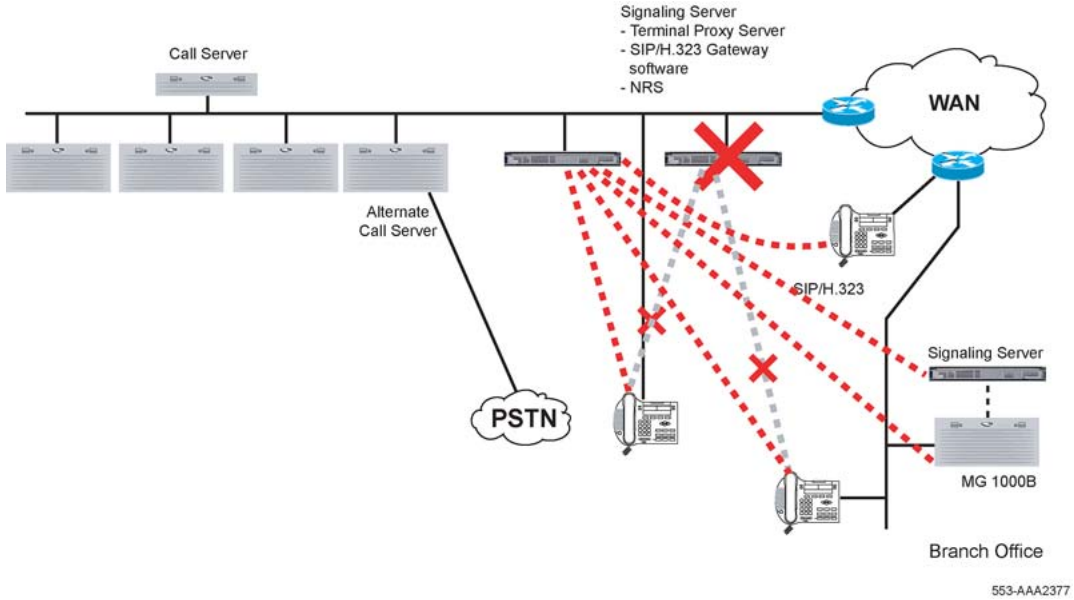

Figure 34: Signaling Server failure

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 34: Signaling Server failure on page 111, two Signaling Servers can load-share when the Media Gateway contain multiple 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.

NRS failure

Figure 35: NRS failure on page 112 illustrates an NRS failure.

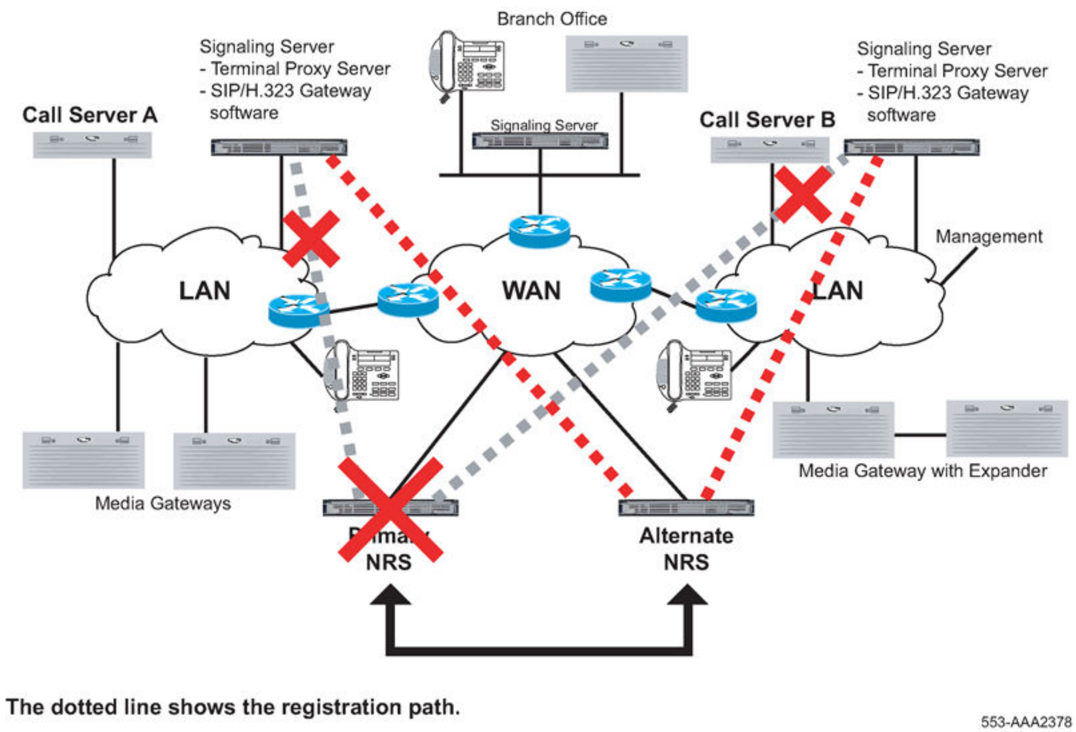

Figure 35: NRS failure

NRS redundancy

Figure 35: NRS failure on page 112 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 36: NRS failure fail-safe on page 113 illustrates NRS fail-safe.

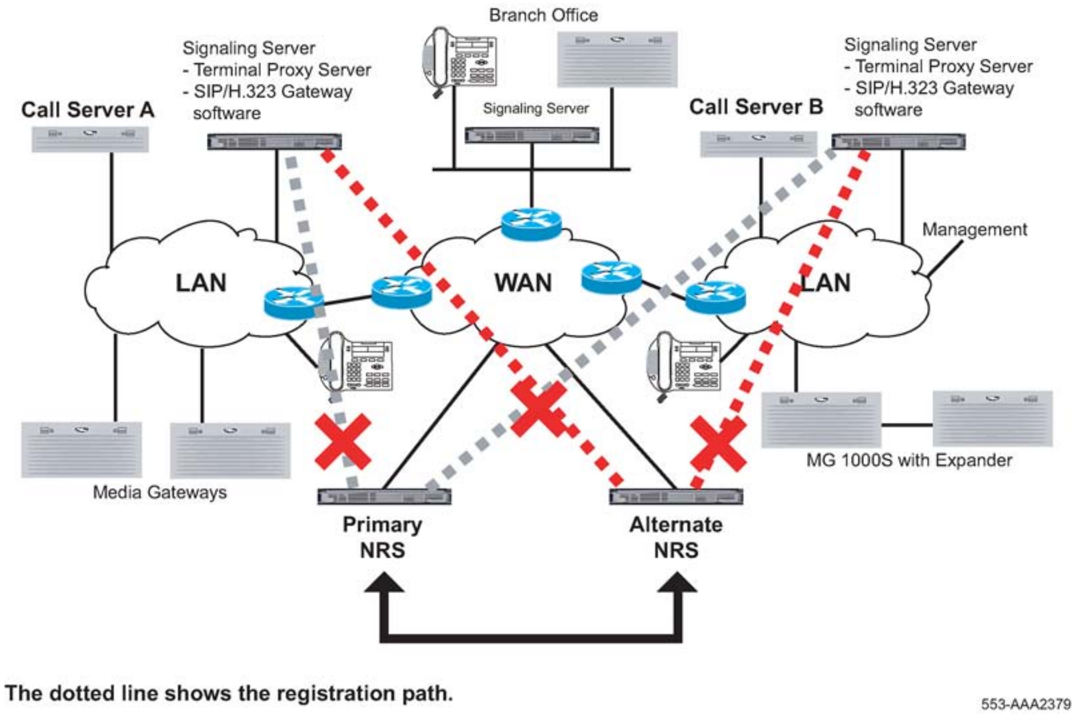

Figure 36: NRS failure fail-safe

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.

CS 1000E resiliency scenarios

This section describes the following resiliency scenarios:

- Call Server failure on page 114 (see Call Server failure on page 114)
- Signaling Server failure on page 116 (see Signaling Server failure on page 116)
- NRS failure on page 117 (see NRS failure on page 117)
- Branch Office scenarios on page 119 (see Branch Office scenarios on page 119)

Refer toFigure 37: CS 1000E on page 114 when reviewing these scenarios.

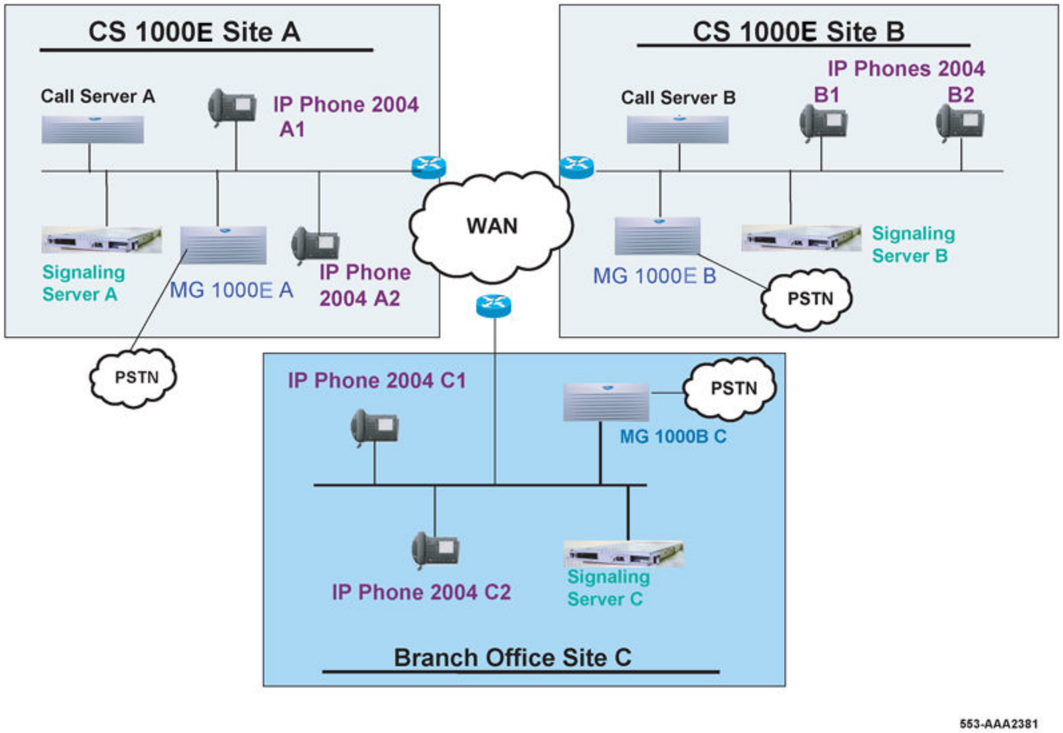

Figure 37: CS 1000E

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 Media Gateway is finished rebooting, and then the call is dropped.

Describe what happens:

Media Gateway 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 Media Gateway 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 Media Gateway is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 1.

Media Gateway 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 Media Gateway A, and Call Server A fails.

What happens to the call in progress?

Same as Scenario 1.

The call stays up until the Media Gateway is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 1.

Media Gateway 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 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 5

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 4.

- 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 4.

- 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 4.

- 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 6

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 7

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 4.

- 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 8

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 6.

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 9

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 Media Gateway A is finished rebooting, and then it is dropped.

Describe what happens:

Media Gateway A reboots at the Main Office site and acts as an alternate Call Server at Site A. The Branch Office 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 10

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

What happens to the call in progress?

Same as Scenario 9.

The call stays up until Media Gateway A is finished rebooting, and then it is dropped.

Describe what happens:

Same as Scenario 9.

Media Gateway A reboots at the Main Office site and acts as an alternate Call Server at Site A. The Branch Office 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 9.

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

Resiliency Scenario 11

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.

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 12

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. 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). At this stage, 100% of telephones from Site A are registered to the redundant Signaling Server.

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 13

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 Branch Office telephones. It is approximately 6 minutes for 400 telephones.

Resiliency Scenario 14

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 15

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

What happens to the call in progress?

The call stays up until Media Gateway A is finished rebooting, and then it is dropped.

Describe what happens:

Media Gateway A reboots at the Main Office site and acts as an alternate Call Server at Site A. The Branch Office 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 16

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 Branch Office IP to IP calls in progress.

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

Not applicable.

Resiliency Scenario 17

IP Phone 2004 C1 is talking to someone off-net over a PSTN trunk in Branch Office 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 Media Gateway 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 18

A digital telephone in the branch office is talking to someone off-net over a PSTN trunk in Branch Office 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 19

A digital telephone in the Main Office is talking to someone off net over a PSTN trunk in Branch Office 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 Media Gateway A talking to someone off-net over a PSTN trunk in Media Gateway 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 20

A digital telephone in the Main Office Site A is talking to someone off-net over a PSTN trunk in Branch Office 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 Media Gateway A talking to someone off-net over a PSTN trunk in Media Gateway 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 21

A digital telephone in the Main Office Site A is talking to a digital telephone in Branch Office 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 Media Gateway A talking to digital telephone in Media Gateway 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 Branch Office 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 22

A digital telephone in the Main Office Site A is talking to IP Phone C1 in Branch Office C and Signaling Server A (Main Office) fails. There is no redundant Signaling Server installed at 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 reboot and attempt to register to Signaling Server A.

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

IP calls between Site A and Site C are offline and can start once the Signaling Server is fixed.

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. A Media Gateway has two modes of operation:

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

Configure IP telephony nodes and survivable Media Gateway for optimal operational efficiency and reliability.

IP telephony node configuration

An IP telephony node is a grouping of Signaling Servers and Media Cards, regardless of the location of theMedia Cards in Media Gateway. Therefore, several Media Gateways can belong to the same node. Alternately, each Media Gateway 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 Media Gateways with MGC that are configured as survivable.

The survivable Media Gateway (alternate Call Server) IP address must be on the same ELAN subnet as the Call Server. If the Media Gateway 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, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

If there are different nodes in different Media Gateway, 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 a Media Gateway 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 Media Gateway.

For a description of the enhanced redundancy for IP Line nodes, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125* .

Alternate Call Server considerations

The following are alternate Call Server considerations:

- Media Gateway are collocated.
- Configure one IP telephony node for the system (that is, all Media Gateway).
- Only one IP Phone Connect Server on a Signaling Server is required for the node.
- Trunks in any Media Gateway can be used by all users.
- DSP resources on Media Cards in any Media Gateway can be used by all users.
- Configure one survivable Media Gateway 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 Media Gateways.
- 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 Media Gateways, but all their local telephones and trunks are functional.
- Less administration is required since there is only one node to manage.

Campus survivable Media Gateway considerations

The following are campus survivable Media Gateway considerations:

- Media Gateways are in different locations.
- Configure a separate IP telephony node for each Media Gateway.
- Each Media Gateway requires an IP Phone Connect Server on the Signaling Server.
- At each Media Gateway, provision trunks to distribute traffic and for survivability.
- At each Media Gateway, provision DSP resources on the Media Cards.
- Configure each survivable Media Gateway 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 Media Gateway.
- 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 Media Gateways, but all their local telephones and trunks are functional.
- More administration is required since there is more than one node to manage.

Multiple D-channels

Avaya does not recommend you to split the Primary and Backup D-channels of the same ISDN Trunk Group across multiple GR/CR CS 1000E Media or PRI Gateways. While this configuration insures D-channel redundancy during some Primary D-Channel failure situations, states could exist where both D-channels register to different Call Servers and simultaneously activate creating a conflict in the Central Office. This conflict can affect service and can lead to a complete ISDN Trunk Group outage in most service provider Central Offices.

If your service provider supports ISDN Trunk Group hunting, Avaya recommends you to maintain multiple ISDN Trunk Groups with each ISDN service provider. Configure each Trunk Group with its own Primary and Backup D-channels on PRI circuits in each Media Gateway. This solution offers resilient configuration in larger systems distributed geographically and operates well even if your service provider is unable to support a D-channel for each ISDN PRI circuit.

Avaya can provide VoIP Session Border Controllers as an alternative to large scale ISDN Trunking facilities. This solution offers improved flexibility in deployment and resiliency performance. For more information, see www.avaya.com/support.

Configuring for survivability

For more information about configuring survivability. see *Avaya System Redundancy Fundamentals, NN43001-507*.

Chapter 9: Telephony planning

Contents

This chapter contains the following topics:

Installation planning

Use Table 7: installation planning on page 129 as a guide to prepare a detailed plan for every installation.

Table 7: installation planning

Procedure	Requirements
Research	Determine requirements for fire protection and safety, the equipment room, grounding and power, and cables.
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 8: Milestone chart on page 130 lists typical activities in a milestone chart. For a complex site, a more detailed chart can be required.

Table 8: 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 an Avaya Communication Server 1000E (Avaya CS 1000E) system, you must evaluate the existing Telecom infrastructure. This evaluation helps decide whether to replace existing network components or add new Avaya 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:
    - *Avaya Telephones and Consoles Fundamentals, NN43001-567*
    - *Avaya IP Phones Fundamentals, NN43001-368*
- System features. For information about feature operation, see *Avaya Feature Listing Reference , NN43001-111*. For information about feature configuration, see *Avaya Software Input Output Administration , NN43001-611*.
- System interworking and networking. For information about Numbering and Dial Plan Configuration, see the following:
    - *Avaya Electronic Switched Network Signaling and Transmission Guidelines, NN43001-280*
    - *Avaya Dialing Plans Reference, NN43001-283*
    - *Avaya Basic Network Feature Fundamentals, NN43001-579*
- PRI/DTI clocking. For information about PRI/DTI clocking, see the following:
    - DTI/PRI clocking on page 133
    - *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*
    - *Avaya ISDN Basic Rate Interface Feature Fundamentals, NN43001-580*

Applications

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

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

Access

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

- *Avaya ISDN Primary Rate Interface Fundamentals, NN43001-569*
- *Avaya ISDN Basic Rate Interface Feature Fundamentals, NN43001-580*

For information about FXS, FXO, or ground/loop start COT trunks, see *Avaya 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)
- Zone Based Dialing (ZBD)
- Transferable Directory Numbers (TNDN)

See *Avaya Network Routing Service Fundamentals, NN43001-130* 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, see *Avaya 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 38: Hierarchical synchronization on page 134.

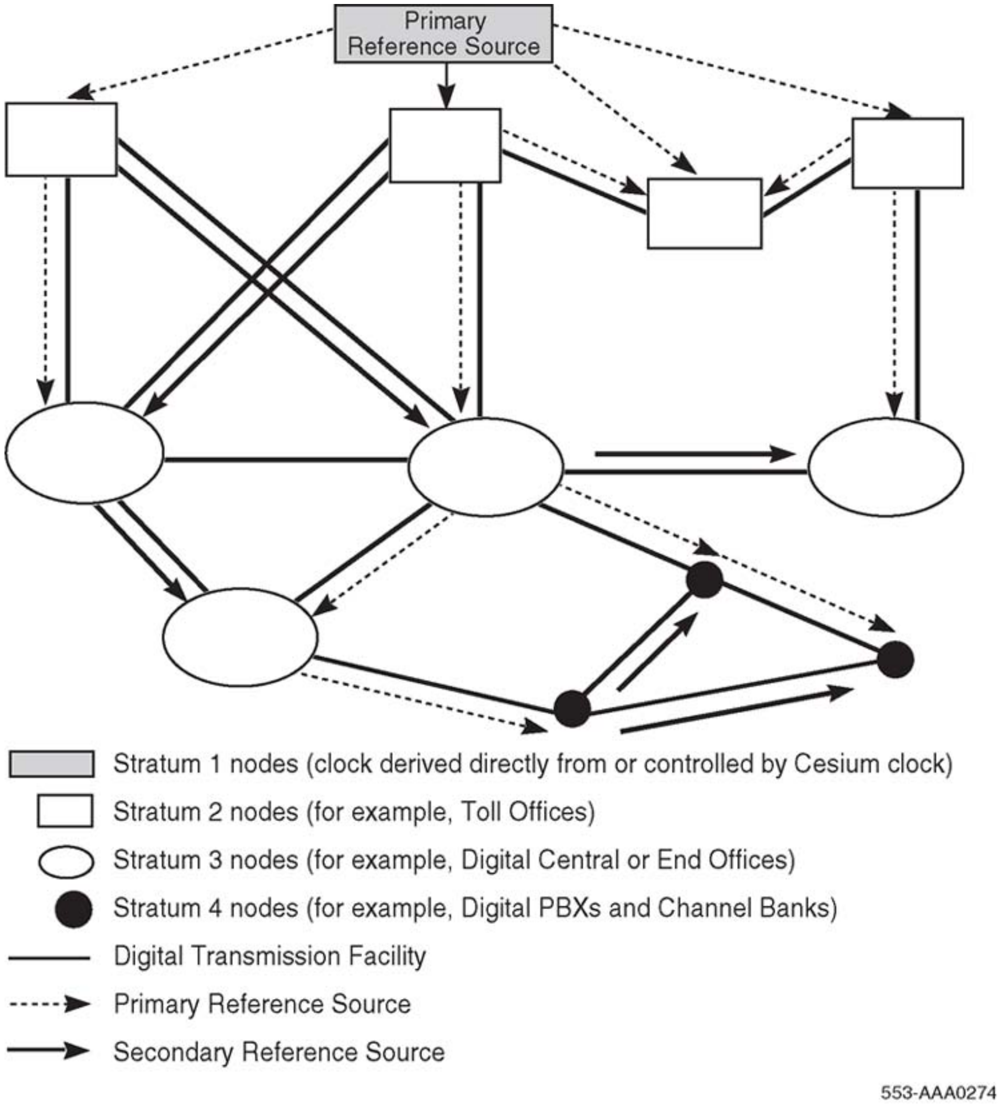

Figure 38: Hierarchical synchronization

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, Avaya 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 38: Hierarchical synchronization on page 134 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 38: Hierarchical synchronization on page 134 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 10: Node categories on page 137, 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 9: 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 Duplication	Required	Required Non-duplicated hardware that meets all other Stratum 3 requirements is referred to as Stratum 3ND.	Not Required
Maximum Time Interval Error (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.
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 10: 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 a Media Gateway 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 Media Gateway.
- 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 Media Gateway Expander does not support clock controllers.

Clock controller function and description

The NTAK20 A-series clock controller meets Stratum 3 level requirements and the NTAK20 B-series clock controller meets Stratum 4 requirements. The embedded clock controllers on the NTAK10 2MB DTI card, and the NTAK79 2MB PRI card meet Stratum level 4 requirements.

Clocking modes

The CS 1000E system supports up to one clock controller in each Media Gateway. Each clock controller can operate in one of two modes: tracking or nontracking (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 Media Gateway) 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 tracking 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 autorecovery.

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.

Automatic 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 (nontracking)

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 11: NTAK20 LED indications on page 140 provides a description of the NTAK20 LEDs.

Table 11: 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.

LED state	LED color	Definition
		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, a Media Gateway can support only one clock controller, and a Media Gateway Expander cannot support a clock controller.

The following are clock controller acronyms:

- CC - Clock Controller
- FRUN - Free Running mode
- PREF - Primary Reference
- SREF - Secondary Reference

Free-running clocks

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

Figure 39: Acceptable connection to an isolated private network with primary reference on page 141 to Figure 41: Acceptable connection with a combined CO and private network on page 142 show acceptable connections.

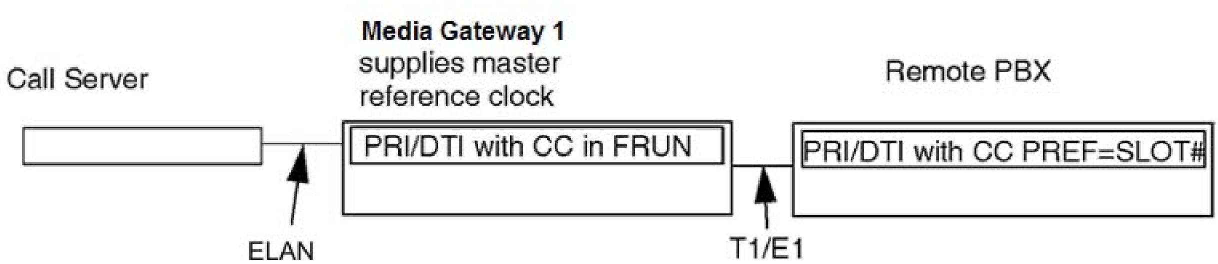

Figure 39: Acceptable connection to an isolated private network with primary reference

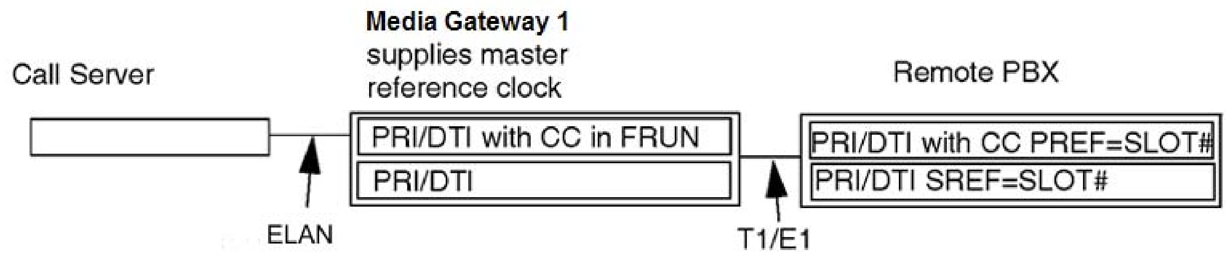

Figure 40: Acceptable connection to an isolated private system with primary and secondary reference

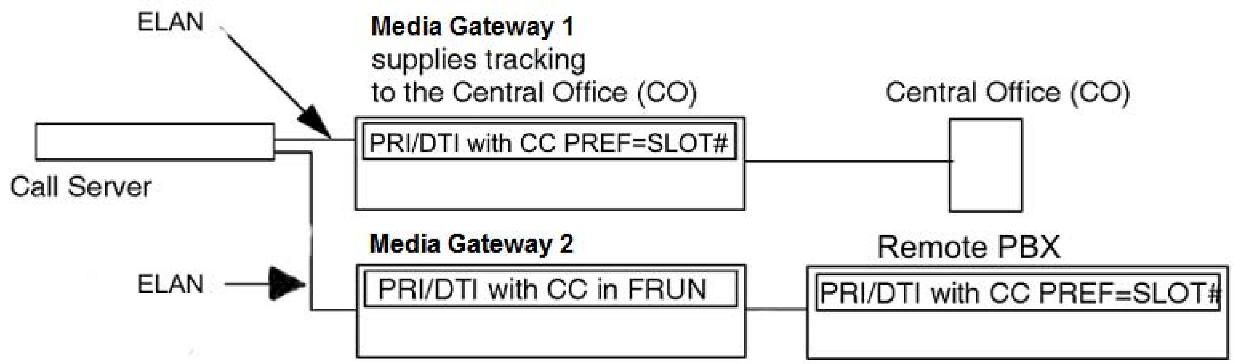

Figure 41: Acceptable connection with a combined CO and private network

Connecting to a CO

Any Media Gateway 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.

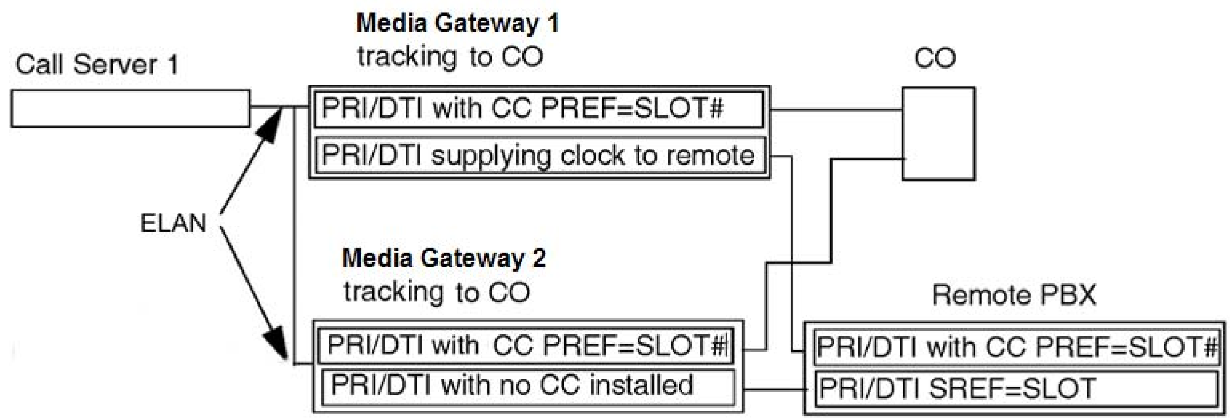

Figure 42: Acceptable connection: Media Gateway 1 and Media Gateway 2 receive clock reference directly from CO

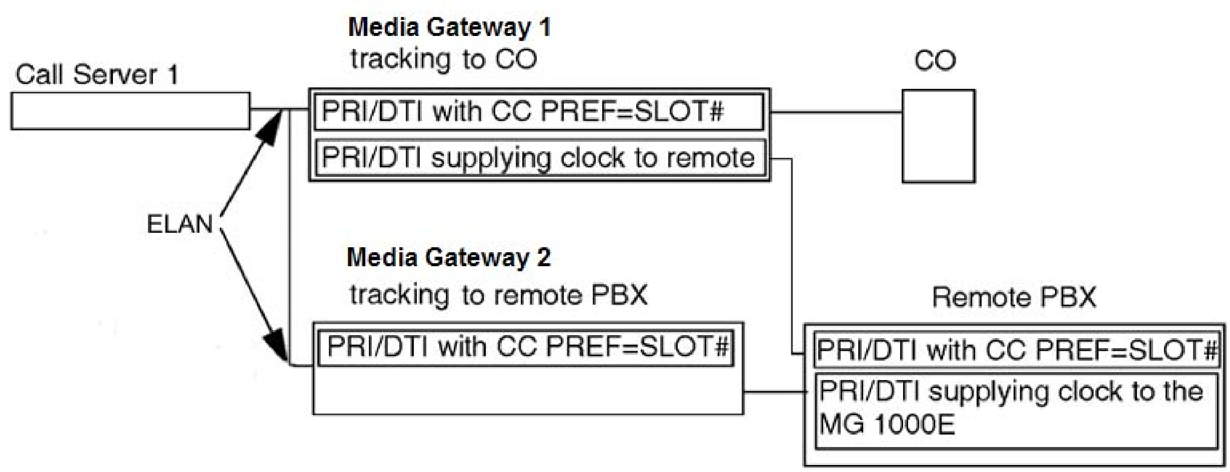

Figure 43: Acceptable connection: Media Gateway 1 receives clock reference directly from CO/ Remote, Media Gateway 2 receives clock reference indirectly from CO

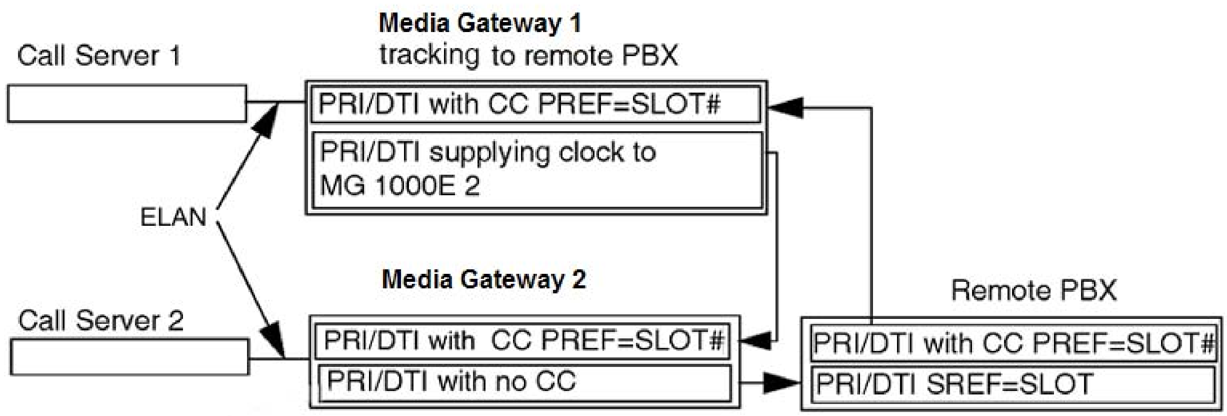

Figure 44: Unacceptable connection: Media Gateway 1 references remote PBX; clock loop, no master clock reference

Allocating primary and secondary references

The secondary reference (SREF) clock must reside in the Media Gateway with the primary (PREF) reference.

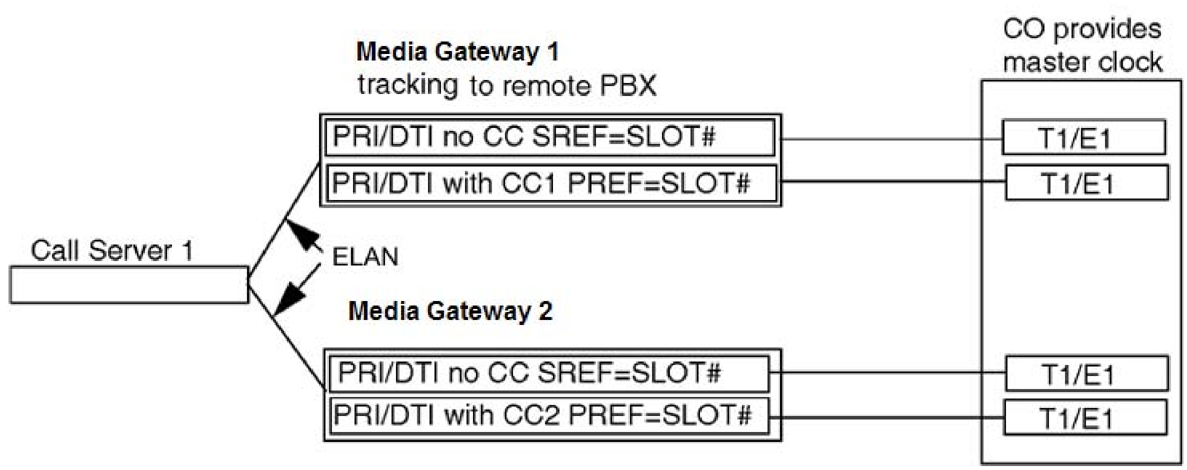

Figure 45: Acceptable connection: Media Gateway 1 references remote PBX; Media Gateway 2 provides master reference to remote PBX

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 46: NTAK20 Daughterboard installation on page 145 shows the installation of the NTAK20 on the NTRB21 TMDI card.

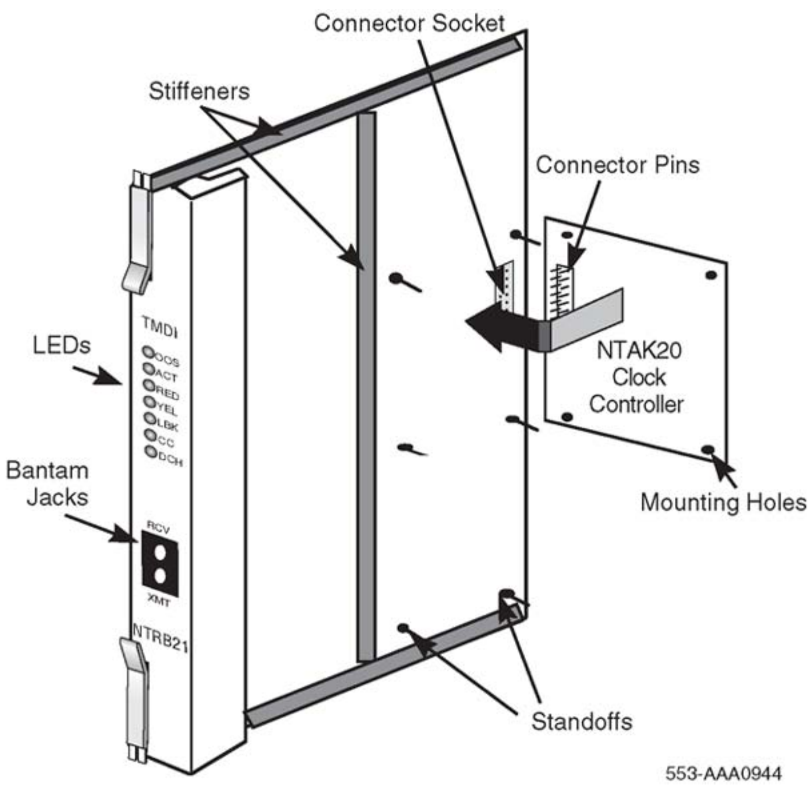

Figure 46: NTAK20 Daughterboard installation

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

Table 12: 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
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.

Table 13: 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 Media Gateway 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:

Table 14: 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.

Table 15: 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

Command	Description
	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 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 Media Gateway with the following command.

Table 16: 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.

Chapter 10: Preparing a system installation plan

Contents

This chapter contains the following topics:

Introduction

Warning:

Before an Avaya Communication Server 1000E (Avaya 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 about the minimum VoIP network requirements and converging a data network with VoIP, see *Avaya 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 17: Installation plan outline on page 151 as a guide for preparing a detailed installation plan.

Table 17: 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 18: Milestone chart on page 151 lists typical activities included in a milestone chart. For a complex site, you must create a more detailed chart.

Table 18: 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.
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.

Task	Action
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:**

Avaya 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.

Important:

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.

Important:

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 Avaya 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.
- The room must meet the requirements for temperature and humidity. For more information about temperature and humidity requirements, see Temperature and humidity control on page 156 and Air conditioning guidelines on page 158.

- The room cooling system must meet the requirements for the installed equipment. For estimating cooling requirements based on thermal generation from system components, see Power consumption on page 185.
- Select a location for equipment installation that is not subject to constant vibration.
- Locate equipment at least 12 ft (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.

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 19: Operating environment on page 157 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 19: Operating environment

Equipment	Temperature and humidity considerations
CS 1000E	Recommended: • 15° to 30°C (59° to 86°F) • RH 20% to 55%, noncondensing Absolute: • 10° to 45° C (50° to 113°F) • RH 20% to 80%, noncondensing • temperature change less than 10°C (18°F) per hour
COTS Servers (IBM, Dell, HP)	Recommended: • 10°C to 35°C (50° to 95°F) • RH 20% to 80%, noncondensing
Telephones	Absolute: • 5°C to 40°C (41° to 104°F) • RH 5% to 95%, noncondensing
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 20: Storage environment on page 157 to store or transport equipment.

Table 20: Storage environment

Equipment	Temperature/humidity considerations
CS 1000E system	• –40° to 70°C (–40° to 158°F) • RH 5% to 95%, noncondensing
COTS servers (IBM, Dell, HP)	• –40° to 60°C (–40° to 140°F) • RH 8% to 80%, noncondensing
Telephones	• –40° to 70°C (–40° to 158°F) • RH 5% to 95%, noncondensing

Equipment	Temperature/humidity considerations
Media Gateways	• –40° to 70°C (–40° to 158°F) • RH 5% to 95%, noncondensing
Other terminal devices	Refer to the specific Avaya publication or the manufacturer's guidelines
Important: 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 23: Current, power, and cooling requirements for CS 1000E components on page 185 and Table 24: Power and cooling requirements for Media Gateway packs on page 186 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.

Important:

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. Avaya 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

For more information, see Power and grounding on page 173 .

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 1000E. The cable also functions as a remote TTY if it has been configured with an MGC. On the Avaya MG 1000E, 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.
- Surge-suppression cables: These cables prevent transient voltages from damaging certain Central Office Trunk (COT) and Direct-Dial Inward (DDI) cards. The cable has a male connector on one end and a female connector on the other so that you can connect it serially with the existing cable. For a list of cards that require surge-supression cables and installation instructions, see *Circuit Card Reference, NN43001–311*.

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 Media Gateway. 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 Media Gateway provide connectivity.
- Ensure that the 100BaseTx Layer 2 (or Layer 3) switch supports full-duplex connection. Routers are not supported in Call Server to Media Gateway 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 about the requirements for creating a robust, redundant network, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Keep a record of the IP addresses assigned to system components. See Figure 47: Sample IP address record sheet on page 164 for a sample.

ELAN Subnet Number	MAC address (by byte/octet) B1	B2	B3	B4	B5	B6	Equipment Description	Serial Number	Comment	TLAN Subnet Address	Location Rack	L/S/C
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												
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	NNTMG19XK Vxx	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	NNTMG19XK Vxx	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	NNTMG19XK Vxx	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	NNTMG19XK Vxx	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	NNTMG19XK Vxx	MGE 5		R2	8/0
xx.xx.xxx.173	00	00	75	xx	xx	xx	SSC-6	NNTMG19XK Vxx	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/M
xx.xx.xxx.205	00	20	D8	xx	xx	xx	Media Card	NNTMEJ02Bxxx	MGE 6	xxx.xx.x.137	R2	8/1/M
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												

Figure 47: Sample IP address record sheet

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

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 48: Building cable zones on page 168 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 on page 160 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 49: Sample cable record on page 169 for an example.

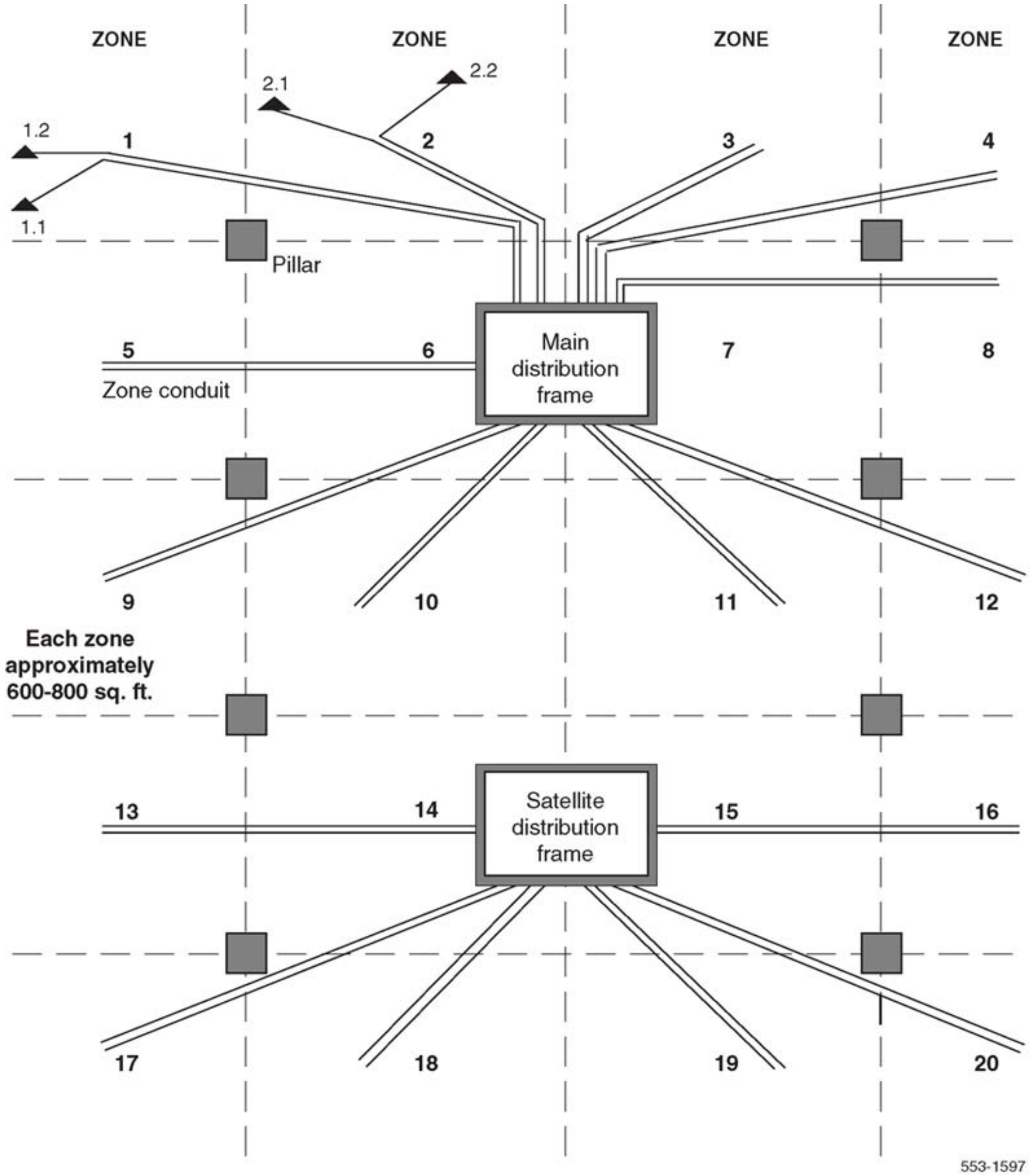

Figure 48: Building cable zones

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

553-1595

Figure 49: Sample cable record

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 publications, are delivered with each system.

Chapter 11: Power and grounding

Contents

This chapter contains the following topics:

Introduction

Avaya Communication Server 1000E (Avaya 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, Avaya 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, Avaya 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.

Voltage:

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 21: Area-specific ground wire requirements on page 174 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 21: 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).

Voltage:

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 Avaya standard. Table 22: Grounding design considerations on page 175 describes grounding design considerations.

Table 22: 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

Voltage:

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 51: Typical wiring plan on page 178.

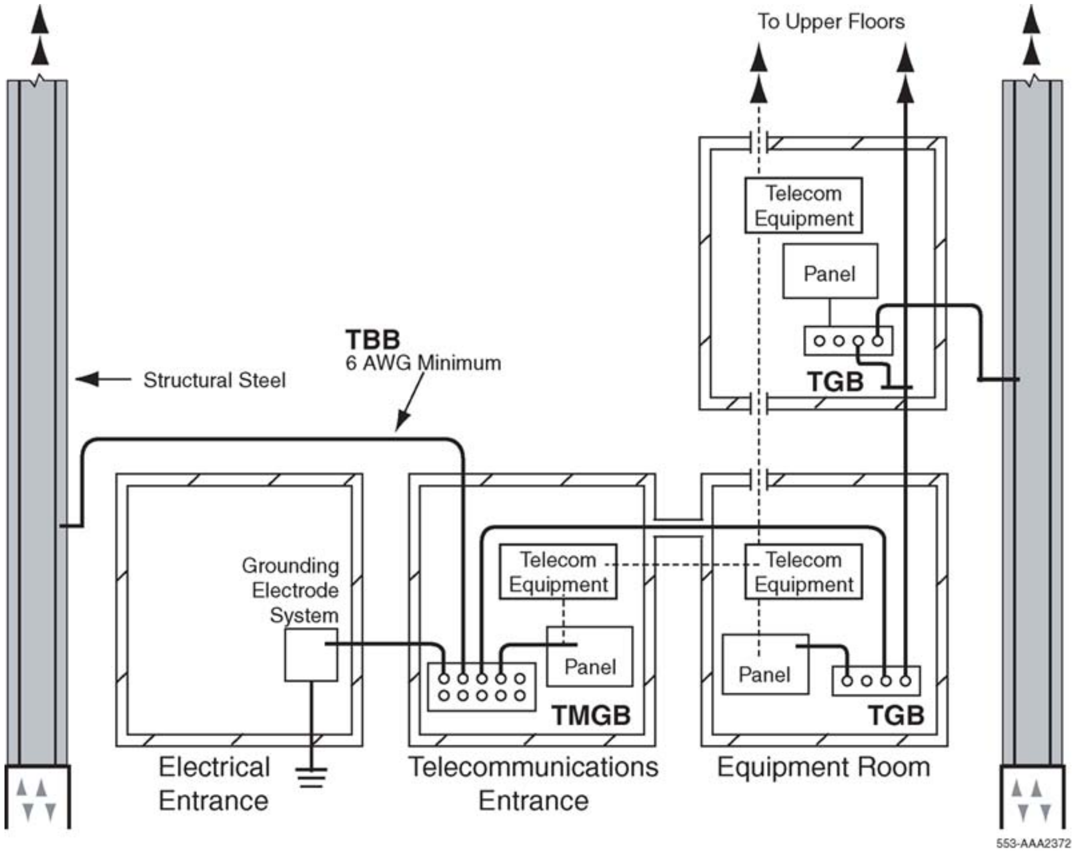

Figure 50: ANSI/TIA/EIA-607 grounding schematic

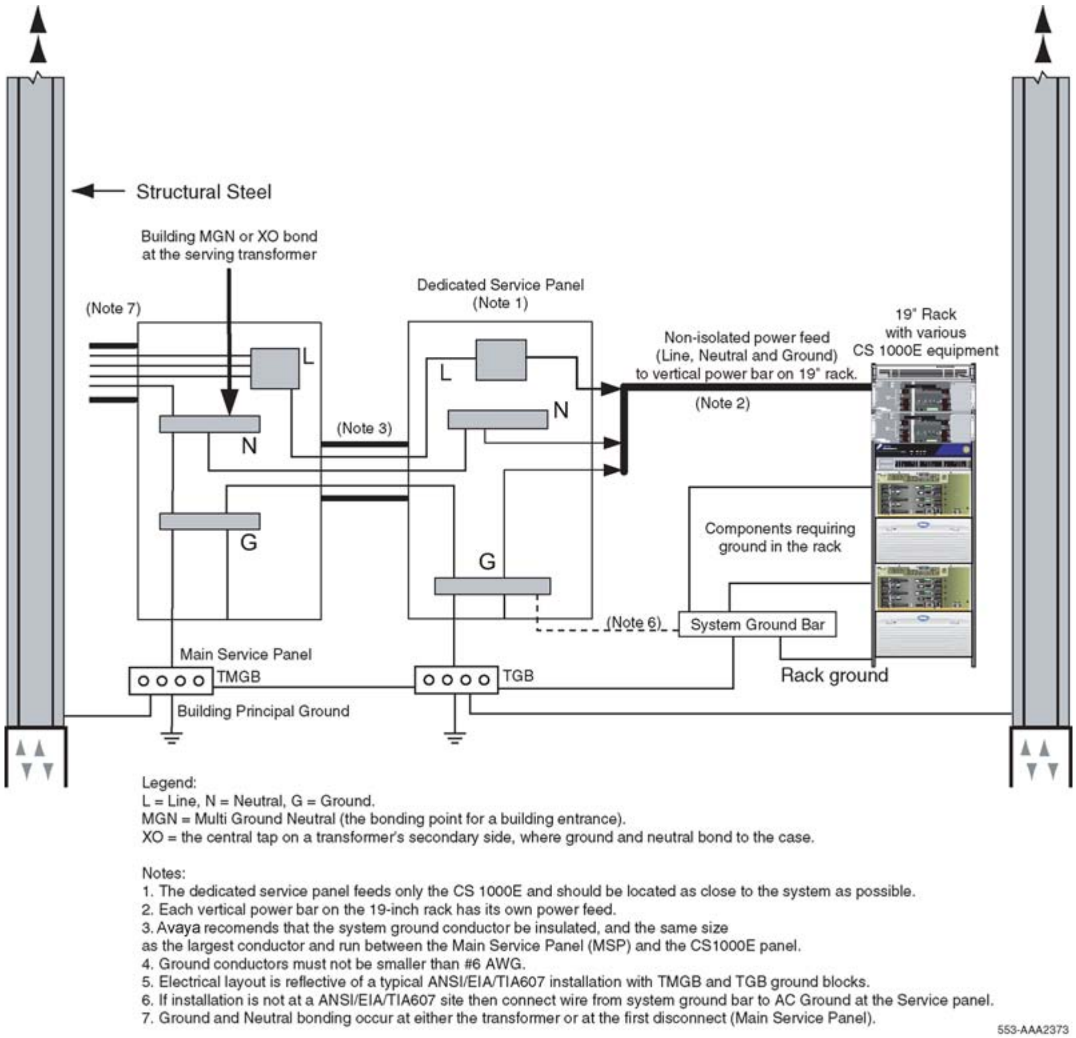

Figure 51: Typical wiring plan

Grounding methods

This section describes the grounding methods for:

- Ground bar (NTBK80) on page 179
- Ground bar (NTDU6201) on page 179
- CP PIV Call Server (NTDU62) on page 179
- COTS servers on page 180
- Media Gateway on page 180

- Media Gateway Expander on page 180
- Multiple components in a rack on page 181

Voltage:

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 21: Area-specific ground wire requirements on page 174 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 51: Typical wiring plan on page 178.

CP PIV Call Server (NTDU62)

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

- The CP PIV Call 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 CP PIV Call Server power cord is plugged into a wall AC outlet. The CP PIV Call Server is grounded outside of the rack via the safety grounding conductor in the power cord. This method only ensures proper grounding of the CP PIV Call Server. It does not provide grounding protection for other rack-mounted pieces of equipment in a CS 1000E system. Therefore, ensure that other devices in the rack are properly grounded as required.

The CP PM Call Server is grounded from the Media Gateway

COTS servers

The Commercial off-the-shelf (COTS) server does not connect to a ground bar. It is properly grounded when:

- The COTS 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 COTS server power cord is plugged into a wall AC outlet. The 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 21: Area-specific ground wire requirements on page 174 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.

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 1000E and Media Gateway Expander

Each Media Gateway 1000E Expander pair must share the same electrical breaker and outlet. For more information, see AC input requirements on page 184.

If the system is equipped with Avaya CallPilot, the CallPilot server must connect to the same dedicated service panel that feeds the Media Gateway 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 nonisolated ground type.

Powering redundant equipment

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

For redundant power in the Media Gateway 1010, each of the two MG 1010 power supplies must be on separate circuits.

Powering auxiliary equipment

Terminals, printers, modems, and other data units used with the CS 1000E or Media Gateway 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 nonauthorized 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. To determine the system power consumption, see Table 23:

Current, power, and cooling requirements for CS 1000E components on page 185 and Table 24: Power and cooling requirements for Media Gateway packs on page 186.

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.

⚠ **Voltage:**

DANGER OF ELECTRIC SHOCK

Avaya 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.

Installing an isolation transformer without pluggable power cords on page 183 describes the method to install an isolation transformer without pluggable power cords.

Installing an isolation transformer without pluggable power cords

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 52: Typical hardwired isolation transformer wiring plan on page 184 shows the isolation transformer connections.

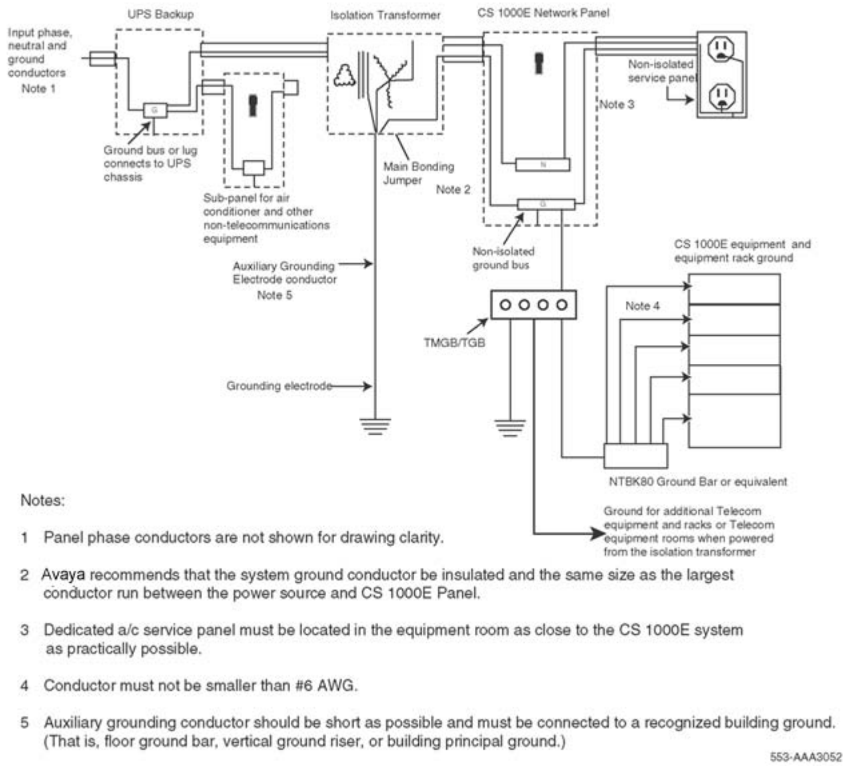

Figure 52: Typical hardwired isolation transformer wiring plan

AC input requirements

For the AC input current requirements of Communication Server 1000E components, see Table 23: Current, power, and cooling requirements for CS 1000E components on page 185 .

North America: Voltage range 90 to 132V AC, 60Hz.

Europe and UK: Voltage range 180 to 250V AC, 50Hz Note: Regulations in Germany allow a maximum supply panel fuse or breaker of 16A.

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 23: Current, power, and cooling requirements for CS 1000E components on page 185 summarizes the current, power, and cooling requirements for CS 1000E components. Table 23: Current, power, and cooling requirements for CS 1000E components on page 185 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 23: Current, power, and cooling requirements for CS 1000E components

Component	Current @ 120/240 V AC (A)		Required UPS power (W)		Thermal dissipation (Btu)	
	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
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
NTDW40 (IBM x3350)	4.00/2.0	2.00/1.0	400.00	250.00	1365.00	853.00
NTDW41 (Dell R300)	4.00/2.0	2.00/1.0	400.00	250.00	1365.00	853.00
700501181 (HP DL360 G7)	4.00/2.0	2.00/1.0	400.00	250.00	1365.00	853.00
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
Ethernet Routing Switch 2526T	0.8	-	96.00	76.00	327.00	-
Ethernet Routing Switch 2550T	0.8	-	96.00	76.00	327.00	-

Component	Current @ 120/240 V AC (A)		Required UPS power (W)		Thermal dissipation (Btu)	
	Maximum	Typical	Maximum	Typical	Maximum	Typical
Ethernet Routing Switch 2526T-PWR	2.9	-	350.00	280.00	327.00	-
Ethernet Routing Switch 2550T-PWR	2.9	-	350.00	280.00	580.00	-
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. Avaya recommends that qualified personnel take current measurements for a more accurate assessment of UPS and thermal requirements. Maximum AC input for the Ethernet Routing Switch 2550T-PWR and the Ethernet Routing Switch 2526T-PWR 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 24: Power and cooling requirements for Media Gateway packs on page 186 provides the power consumption and thermal dissipation of Media Gateway packs (circuit cards and daughterboards) commonly installed in CS 1000E and Media Gateway Media Gateways and Expanders. Use the data in the following table in conjunction with the system and Media Gateway power consumption worksheets. See Power consumption worksheets on page 188.

Electrical load for analog line cards varies with traffic load. The figures in the following table assume that 50% of analog lines are active.

For digital and analog (500/2500-type) telephones, most thermal dissipation will be external to the switch room. This is accounted for In Table 24: Power and cooling requirements for Media Gateway packs on page 186, and the Power consumption worksheets on page 188. This thermal dissipation is also accounted for in the typical values shown in Table 23: Current, power, and cooling requirements for CS 1000E components on page 185.

Table 24: Power and cooling requirements for Media Gateway packs

Component	Active off-hook (%)	Power consumption (W)	Thermal dissipation outside system (W)
NTAK11 Option 11C Cabinet (NTDK70 power supply)	N/A	8.5	
NTAK14/15 MG 1000E Chassis	N/A	24	

Component	Active off-hook (%)	Power consumption (W)	Thermal dissipation outside system (W)
NTC310 MG 1010 Chassis (2 power supplies, 3 blower fans, 1 MGU card)	N/A	100	
NTDW53/NTDW54 Common Processor Dual Core	N/A	30	
NTDW56/NTDW59 Common Processor Media Gateway	N/A	35	
NTDW60/98 Media Gateway Controller card	N/A	7	
NTDW61/66/99 Common Processor Pentium Mobile	N/A	30	
NTDW62 MGC DSP daughterboard (32 port)	N/A	4	
NTDW64 MGC DSP daughterboard (64 port)	N/A	4	
NTDW78 MGC DSP daughterboard (128 port)	N/A	4	
NT5K02 Flexible Analog Line card	50	26	19.4
NT8D02 Digital Line card	100	26	13
NT8D03 Analog Line card	50	26	19.4
NT8D09 Analog Message Waiting Line card	50	26	19.4
NT8D14 Universal Trunk card (DID enabled)	N/A	28	
NT8D15 E&M Trunk card	N/A	29	
NTAK09 1.5 MB DTI/PRI card	N/A	10	
NTAK10 2.0 MB DTI card	N/A	12	
NTAK79 2.0 MB PRI card	N/A	12	
NTBK50 2.0 MB PRI card	N/A	12	
NTRB21 TMDI (1.5 MB DTI/PRI) card	N/A	12	
NTVQ01 Media Card (32 port)	N/A	18	
NTDW65 MC32S Media Card (32 port)	N/A	9	

Component	Active off-hook (%)	Power consumption (W)	Thermal dissipation outside system (W)
NTDW79 UDT	N/A	12	

Important:

To determine the required UPS rating for Media Gateways you must allow for the efficiency factor of the Media Gateway power supply plus peak inrush. For NTAK11, NTDU14, and NTDU15, multiply the total power consumption of the components by 1.5. For NTC310, multiply the total power consumption of the components by 1.3. See the Media Gateway power consumption worksheets for this calculation.

Power consumption worksheets

Use the provided worksheets to determine power consumption for the CS 1000E system.

- Table 25: CS 1000E system power consumption worksheet on page 188
- Table 26: NTAK11 Media Gateway Option 11C power consumption worksheet on page 189
- Table 27: NTDU14 and NTDU15 Media Gateway power consumption worksheet on page 190
- Table 28: NTC310 Media Gateway 1010 power consumption worksheet on page 191

Prepare one worksheet for the system as a whole (Table 25: CS 1000E system power consumption worksheet on page 188).

Table 25: 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					
Ethernet Routing Switch 2526T					
Ethernet Routing Switch 2550T					

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)
Ethernet Routing Switch 2526T-PWR					
Ethernet Routing Switch 2550T-PWR					
NTDW77 HD PRI Gateway					
NTAK11 Media Gateway Option 11C*					
NTDU14/15 Media Gateway 1000E*					
NTC310 Media Gateway 1010*					
TOTAL					
*Enter the sum of the totals from all Media Gateway worksheets.					

Prepare one worksheet for each Media Gateway. Use the appropriate worksheet for the Media Gateway type from the following list.

- Table 26: NTAK11 Media Gateway Option 11C power consumption worksheet on page 189
- Table 27: NTDU14 and NTDU15 Media Gateway power consumption worksheet on page 190
- Table 28: NTC310 Media Gateway 1010 power consumption worksheet on page 191

For the power and thermal dissipation requirements for the individual components, see Table 24: Power and cooling requirements for Media Gateway packs on page 186 .

Table 26: NTAK11 Media Gateway Option 11C power consumption worksheet

Media Gateway number ________________			
Slot	Media Gateway Pack	Power consumption (W)	Thermal dissipation outside system (W)
	NTAK11 Option 11C cabinet	8.5	
0	Gateway Controller		
0	DSP daughterboard		

Media Gateway number ________________			
Slot	**Media Gateway Pack**	**Power consumption (W)**	**Thermal dissipation outside system (W)**
0	DSP daughterboard		
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			
TOTAL			
	multiply total power consumption by 1.5	x 1.5	
Required UPS Power (W or VA)			
	subtract total thermal dissipation outside system (W)		
System thermal dissipation (W)			
	multiply by 3.412 to convert to BTU	x 3.412	
System thermal dissipation (BTU)			

Table 27: NTDU14 and NTDU15 Media Gateway power consumption worksheet

Media Gateway number ________________			
Slot	**Media Gateway Pack**	**Power consumption (W)**	**Thermal dissipation outside system (W)**
	NTDU14 MG 1000E chassis	24	
0	Gateway Controller		
0	DSP daughterboard		
0	DSP daughterboard		

Media Gateway number ________________			
Slot	**Media Gateway Pack**	**Power consumption (W)**	**Thermal dissipation outside system (W)**
1			
2			
3			
4			
	NTDU15 Media Gateway Expander chassis	24	
7			
8			
9			
10			
TOTAL			
	multiply total power consumption by 1.5	x 1.5	
Required UPS Power (W or VA)			
	subtract total thermal dissipation outside system (W)		
System thermal dissipation (W)			
	multiply by 3.412 to convert to BTU	x 3.412	
System thermal dissipation (BTU)			

Table 28: NTC310 Media Gateway 1010 power consumption worksheet

Media Gateway number ________________			
Slot	**Media Gateway Pack**	**Power consumption (W)**	**Thermal dissipation outside system (W)**
23	Server card		
22	Server card		

Media Gateway number ________________			
Slot	**Media Gateway Pack**	**Power consumption (W)**	**Thermal dissipation outside system (W)**
	MG 1010 chassis	100	
0	Gateway Controller		
0	DSP daughterboard		
0	DSP daughterboard		
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			
TOTAL			
	multiply total power consumption by 1.5	x 1.5	
Required UPS Power (W or VA)			
	subtract total thermal dissipation outside system (W)		
System thermal dissipation (W)			
	multiply by 3.412 to convert to BTU	x 3.412	
System thermal dissipation (BTU)			

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.

Use the Power consumption worksheets on page 188 to determine 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 23: Current, power, and cooling requirements for CS 1000E components on page 185 and Table 24: Power and cooling requirements for Media Gateway packs on page 186 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, Enterprise Configurator 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.

$$A_{hr} = \left(\frac{W_{total}}{V_{dischg} \bullet \eta_{eff}}\right) \bullet T_{reserve}$$

UPS installation

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

Avaya 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 53: Typical UPS wiring plan on page 195 shows a typical UPS wiring plan.

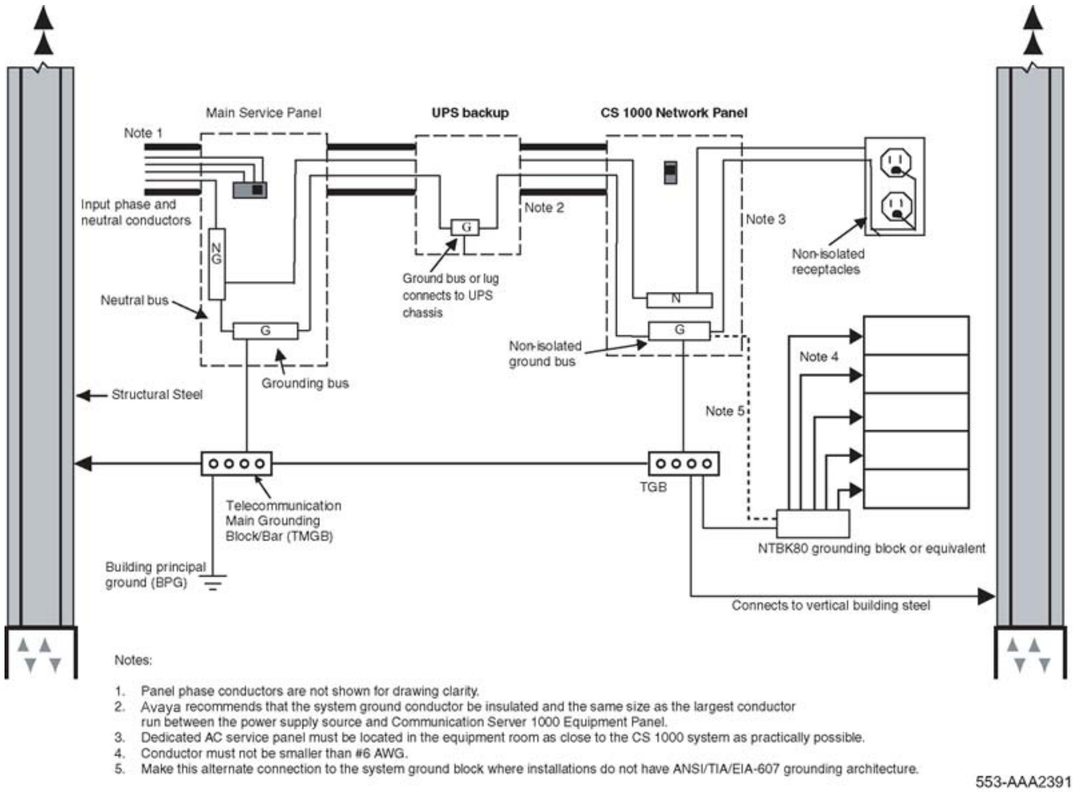

Figure 53: Typical UPS wiring plan

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 Ethernet Routing Switch 2526T-PWR (see Layer 2 switch on page 91). For more information about Power over LAN, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Chapter 12: Design parameters

Contents

This chapter contains the following topics:

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

The term Media Gateway refers to the Avaya CS 1000 Media Gateway 1000E (Avaya MG 1000E), and Media Gateway 1010, (MG 1010). The MG 1010 provides ten IPE slots. The Avaya MG 1000E provides four IPE slots.

Each MG 1000E can be connected to an optional Media Gateway Expander in order to increase capacity to eight IPE slots. In this chapter, the term MG 1000E includes the optional Media Gateway Expander, if equipped.

System parameters

Table 29: System parameters on page 198 lists system parameters and provides their maximum values.

Table 29: 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 – Avaya Communication Server 1000E (Avaya CS 1000E)	65 534	Software design limit. Actual number of TNs will be constrained by physical capacity, real time, memory, and License limits.
Media Cards	256	Limit of Media Cards, which includes banks of 32 DSPs on DSP daughter cards.
DSPs	8192	DSP limit on a system (256 * 32)

Customer parameters

Table 30: Customer parameters on page 198 lists customer parameters and their maximum values.

Table 30: Customer parameters

Customer parameters	Maximum value	Comments
Tenants	512	

Customer parameters	Maximum value	Comments
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 31: Console and telephone related parameters on page 199 lists console and telephone-related parameters and their maximum values.

Table 31: 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: – Avaya 2250 Attendant Console	20	
Incoming call indicators per console	20	
Trunk group busy indicators per console: – Avaya 2250 Attendant Console	20	
Additional key/lamp strips: – console – telephones	2 6	
Add on modules: – Avaya 3904 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 32: Trunk and network-related parameters on page 200 lists trunk and network-related parameters and their maximum values.

Table 32: 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 33: ACD feature parameters on page 201 lists ACD feature parameters and their maximum values.

Table 33: ACD feature parameters

ACD parameters	Maximum value	Comments
ACD DNs and CDNs per customer	2000 (CP PIV, CP PM)	2000 with ACD-F package (411), 1000 with ACD-E package
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	4	4 with ACD-F package (411), otherwise the limit is 2
Number of ACD-ADS customers	5	
Terminals and printers on CCR	8	
Links per VASID	1	

Special feature parameters

Table 34: Non-ACD feature parameters on page 201 lists nonACD feature parameters and their maximum values.

Table 34: 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.
Mobile Extensions per customer	4000	Software design limit. Can be further constrained by PRI requirements.

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 35: Physical capacity/hardware-related parameters on page 203 lists hardware and capacity parameters and their maximum values for TDM conference.

Table 35: Physical capacity/hardware-related parameters

Physical capacity/hardware parameters	Maximum value (loops)	Comments
MGCONF	2	Provide Conf, and MFS functionality for the cards in that Media Gateway. 30 units for each loop.
MGTDS	2	Provide TDS functionality for the cards in that Media Gateway. 30 units for each loop.
Total service and terminal loops	256	Each superloop requires 4 loops.

Communication Server 1000E Co-resident Call Server and Signaling Server design parameters

An Avaya CS 1000E Co-resident Call Server and Signaling Server (Co-res CS and SS) system uses the same architecture and design parameters as the Communication Server 1000E Standard Availability or High Availability system. The co-resident heavy CPU load constrains the number of supported phones and trunks because both the Call Server and Signaling Server applications run on one CPU. For more information about the Co-res CS and SS physical capacity limitations, see Physical capacity on page 211.

Media Cards

A Media Card is a card that provides additional DSP resources for a Media Gateway beyond the DSP resources provided by the Gateway Controller. Media Card is a term for the Media

Card 32-port secure line card, Media Card 32-port line card, and the Media Card 8-port line card.

In the CS 1000E, Media Cards are used primarily for DSP connections between the TDM devices in a Media Gateway and IP circuits.

Media Cards can be assigned to any nonblocking slot other than slot 0. You must provision each Media Gateway with enough DSP ports to support the TDM devices in that Media Gateway.

Call Server memory related parameters

Table 36: Memory related parameters on page 204 lists Call Server memory related parameters and their maximum values.

Table 36: 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
Number of call registers • (recommended)	26 – 65 000 (35 000)
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)

Parameter	Values
	CS 1000E
Call registers for AML input queues (CSQI)	Up to 25% of total call registers (NCR), minimum 20
Call registers for AML output queues (CSQO)	Up to 25% of total call registers (NCR), minimum 20
Auxiliary input queue	20 – the minimum of 25% of total call registers or 255 (default 20)
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 Avaya 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 up to 25% of total call registers (NCR).	

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 36: Memory related parameters on page 204 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 36: Memory related parameters on page 204. 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 25% of NCR). Note that the value of NCR should be increased to account for the requirements of CSQI and CSQO.

Access Restrictions packet logging memory limits

The Access Restrictions feature, also known as the port blocking facility enables a VxWorks firewall to prevent port-based attacks on the CP PIV, MGC, and MC32S. Enabling the port blocking rule list starts performance statistics and logging that requires 384 kB of memory. 64 kB for the rule list, 64 kB of memory to allow for logging, and 256 kB for performance statistics. Memory logging is limited to use a maximum 55 MB of memory.

You can log the packets to the tty or to a first in first out (FIFO) wrap around memory buffer. The logging data can be written to a file. Port blocking logs are independent from the system logs, and store in a portacc.log file on the removable CF card of the CP PIV, and /e partition of the MGC and MC32S.

Chapter 13: System capacities

Contents

This chapter contains the following topics:

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 on page 249 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 on page 323

Memory size

Table 37: Avaya CS 1000 memory requirements on page 208 shows the minimum amount of memory required for Avaya Communication Server 1000 (Avaya CS 1000) software.

Table 37: Avaya CS 1000 memory requirements

System	FMD Flash Drive required	DRAM memory required
CS 1000E SA or HA (CP PIV)	1 GB	512 MB
CS 1000E SA or HA (CP PM)	1 GB	1 GB
CS 1000E CP PM Co-res CS and SS	N/A	2 GB
CS 1000E CP MG Co-res CS and SS	N/A	4 GB
CS 1000E CP DC Co-res CS and SS	N/A	4 GB
CS 1000E COTS2 Co-res CS and SS	N/A	4 GB
CS 1000E Common Server Co-res CS and SS (HP DL360 G7)	N/A	4 GB
CS 1000E CP PM TDM	N/A	2 GB
CP PIV and CP PM VxWorks based SA and HA Call Servers can be upgraded to a maximum of 1 GB DRAM memory. CP DC cards can be upgraded to a maximum of 4 GB DRAM memory.		

Table 38: Recommended call register counts on page 208 shows the call register count recommended for Communication Server 1000 software, so that the system's memory requirements do not exceed the processor's memory capacity.

Table 38: Recommended call register counts

System	Recommended call register count	Memory required (SL-1 words)	Memory required (MB)
CS 1000E HS (CP PM large deployment, see note)	60 000	13 800 000	52.643
CS 1000E (CP PIV and CP PM)	35 000	8 050 000	30.71
CS 1000E Co-res CS and SS	20 000	4 600 000	17.584
Call registers are 230 SL-1 words long. One SL-1 word is 4 bytes.			

System	Recommended call register count	Memory required (SL-1 words)	Memory required (MB)
Note: A large deployment is greater than 11 500 SIP Line users or greater than 22 500 UNIStim users.			

The following table shows the typical memory configured on each platform. Communication Server 1000 Signaling Servers require a minimum of 2 GB memory for the CP PM platform and 4 GB of memory for all other platforms.

Note:

You must upgrade CP DC or CP MG hardware from 2 GB of memory to 4 GB of memory with a Linux Upgrade Kit. The CP PM, IBM 306m and HP DL340 G4 have deployment restrictions (see Signaling Server capacity limits on page 234).

Table 39: Signaling Server memory requirements

Signaling Server platform	Memory (RAM)
CP PM	2 GB
CP DC	4 GB
HP DL360–G4 or IBM x306m	2 GB
Dell R300 or IBM x3350	4 GB
Common Server (HP DL360 G7)	4 GB

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 centi-call seconds (CCS) for each 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 × (1 – p)] × p + (3 × p^2)

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 × (1 – p)] × p + (3 × p^2)

p (penetration factor) = trunking CCS ÷ Total Voice Traffic

Mass storage

The system processor program and data are loaded from a Fixed Media Disk (FMD). Depending on the hardware platform, the FMD can be a hard disk drive (HDD) or Compact Flash (CF) card.

Software installation

Software, customer databases, and PEPS are delivered to the system using a Removable Media Disk (RMD), either CF card or USB, inserted into the Server. An installation process copies the software to the on-board FMD. The software subsequently operates on the Server FMD.

Database backup

The RMD can also be used for customer database backups.

Physical capacity

The following physical capacities are discussed in this section:

- CS 1000E SA and HA physical capacity on page 211
- CS 1000E Co-resident Call Server and Signaling Server physical capacity on page 212
- CS 1000E TDM physical capacity on page 213

CS 1000E SA and HA physical capacity

The CS 1000E SA or HA system (CP PIV or CP PM) supports a maximum of

- 22 500 IP Phones (UNIStim + 2 × SIP Line), where SIP Line = (SipN telephones + Sip3 telephones) and the total number of SIP Line telephones must be <= 11 500
- 50 Media Gateways

- 8000 TDM telephones
- 100 PRI spans

A CS 1000E SA or HA system configured for IP only can support 40 000 IP Phones (UNIStim + 2 × SIP Line), where SIP Line = (SipN telephones + Sip3 telephones) and the total number of SIP Line telephones must be <= 20 000.

A fully expanded CS 1000E system, with 50 MG 1000Es each equipped with an Media Gateway Expander, provides 400 card slots (50 × 8) to support TDM devices and their required DSP resources.

A fully expanded CS 1000E system, with 50 MG 1010s, provides 500 card slots (50 × 10) to support TDM devices and their required DSP resources.

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 about phantom and virtual loops, see the Global Software Licenses chapter in *Avaya Features and Services Fundamentals, NN43001-106*.

For information about loop and card slot usage and requirements for the Media Gateways in the CS 1000E, see Assigning loops and card slots in the Communication Server 1000E on page 379 .

CS 1000E Co-resident Call Server and Signaling Server physical capacity

The CS 1000E Co-res CS and SS supports a maximum of

- 1000 IP Phones (UNIStim + SIP Line), where SIP Line = (SipN telephones + Sip3 telephones) and the total number of SIP Line telephones must be <= 1000
- 5 Media Gateways
- 800 TDM telephones
- 16 PRI spans
- 400 virtual trunks
- 200 ACD agents
- 5 NRS endpoints with up to 20 routing entries
- 5 Branch Offices

Note:

A Co-resident Call Server and Signaling Server with only 2 GB of memory (such as the CP PM card) cannot support all Signaling Server applications and a NRS. In addition to the application restrictions, there are also management function restrictions. Deploying a

Primary UCM, Deployment Manager, EM, NRSM and Subscriber Manager on a server with only 2 GB of memory is not supported.

For recommended deployment options of a 2 GB CP PM Co-res CS and SS, see Signaling Server capacity limits on page 234.

The Communication Server 1000E Co-res CS and SS is supported on various hardware platforms with varying physical capacities. For platform specific CS 1000E Co-res CS and SS capacity information, see *Avaya Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*.

CS 1000E TDM physical capacity

The CS 1000E TDM supports a maximum of

- 5 Media Gateways
- 800 TDM telephones
- 16 PRI spans
- 200 ACD agents

The CS 1000E TDM is supported on the CP PM, CP DC, and CP MG 128 hardware platforms.

The CS 1000E TDM system does not support any IP Phones (UNIStim, SIP Line, or SIP DECT), virtual trunks, or an NRS. For more information about CS 1000E TDM, see *Avaya Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*.

Signaling and data links

The following signaling and data links are discussed in this section:

- Physical links on page 213
- Functional links on page 214

Physical links

There are two types of physical links to consider:

- Serial Data Interface (SDI) on page 214
- Local Area Network (LAN) on page 214

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. Server cards typically have two SDI ports, COM 1 and COM 2. COM 1 (TTY1) must be used for system installation and upgrades.

Local Area Network (LAN)

The system can communicate over the LAN with Ethernet connections through a Network Interface Card (NIC). The CS 1000E LAN contains subnets for the Embedded Local Area Network (ELAN) and the Telephony Local Area Network (TLAN). 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 autonegotiates 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 SDI port to connect to a terminal/computer (TTY) to receive maintenance commands or to print traffic reports, maintenance messages, or CDR records.

ISDN Signaling (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 (30 in Europe based on E1) 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 DCHI 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 40: I/O interface for applications on page 215 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 40: 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
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:

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 Centi-Call Seconds (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

Communication Server 1000E engineering is typically based on measurements you perform in an hour (typical busy hour). This applies to traffic load in CCS and Network CCS.

Loops and superloops

The number of loops and superloops on a CS 1000E is calculated form the number of lines, trunks, cards, and Media Gateways 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, IP Media Services)
- 2 Media Gateways per Superloop (Media Gateway PRI Gateway counts as 1 Media Gateway)
- Gateway Controller 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.
- IP Media Services are IP tone, IP conference, IP attendant consoles, IP recorded announcer, and IP music.
    - IP tone - 30 units for each tone loop
    - IP conference - 30 units for each conference loop
- 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) + ROUNDUP ((2×(SIPN + SIP3 users)) / 1024)
- Loops = ROUNDUP(Conference Ports / 30) + TDS loops (minimum 1 for each MGC) + PRI or DTI cards + (1000E PRI Gateways × 4)
- Total Loops = Loops + IP tone loops + Superloops × 4
- Total Loops > 256 is an error, too many loops being used.
- Total Superloops = ROUNDUP(Total Loops / 4)

Note:

Conference ports can be TDM or IP.

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 nonblocking (one virtual talkslot per virtual TN). This also removes the CCS per superloop constraint.

The Media Gateway has a nonblock 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 Media Gateway as the phone requiring the resource.

Loop capacity and Media Gateway TDM resources are subject to the Grade-of-Service (GoS) described under Grade-of-Service on page 225.

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 54: CS 1000E system call types on page 219 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.

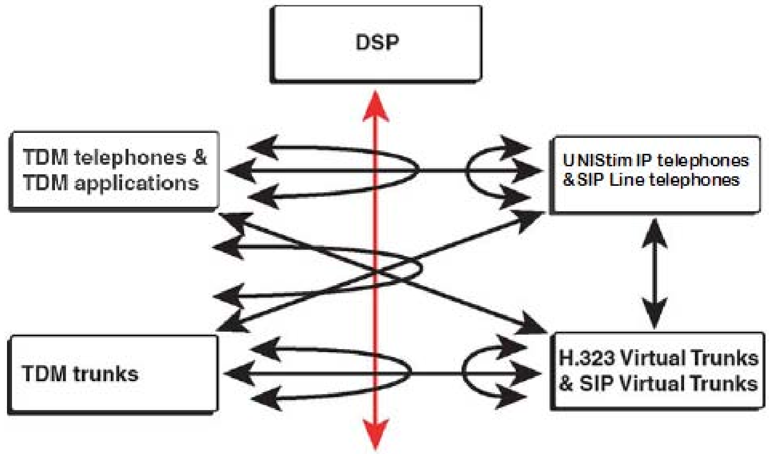

Figure 54: CS 1000E system call types

Table 41: Connection type resources required on page 219 lists the resources required for each type of connection.

Table 41: Connection type resources required

Connection Type	Resources
TDM to UNIStim IP, UNIStim IP to TDM	DSP
TDM to VT, VT to TDM	DSP and VT
UNIStim IP to UNIStim IP	no DSP, no VT
UNIStim IP to VT or VT to UNIStim IP	VT
TDM to TDM telephone or trunk calls	no DSP*, no VT
TDM to SIP Line or SIP Line to TDM	DSP, VT
SIP Line to SIP Line	VT
SIP Line to VT or VT to SIP Line	VT
SIP Line to UNIStim IP or UNIStim IP to SIP Line	VT
* You require DSPs in each Media Gateway for TDM calls between Media Gateway chassis.	

See Resource calculations on page 249 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 (MGTDS and MGCONF) must be defined for use by each Media Gateway.

This section describes the traffic characteristics, calculation algorithms, and impact on other system resources of the following types of service circuits:

- TDS on page 220
- Conference on page 220
- Broadcast circuits on page 220
- DTR on page 221
- IP Media Services on page 222

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 Media Gateway. The TDS circuits are provided by the MGC card. If additional TDS circuits are required in any Media Gateway, a second TDS loop can be configured in it. TDS circuits in a Media Gateway provide tones for TDM telephones or trunks in that Media Gateway 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 Media Gateway. The maximum number of parties involved in a single conference on a Media Gateway with Gateway Controller is 30. Conference circuits in the CS 1000E are a system resource. Conference loops can be TDM or IP.

Broadcast circuits

The Avaya 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 a Media Gateway. 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 on page 387).

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 Media Gateway that they reside.

The MGC 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:

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 on page 422 to determine the number of DTRs required. Note that a weighted average for holding times should be used.

IP Media Services

IP Media Services is a term used for a group of services. The services supported are:

- IP Tone generation for IP Phones
- IP Conference
- IP Music
- IP Recorded Announcer
- IP Attendant Console

IP Tone

IP Tone is required for IP Phone blind transfers (ring back tone required). Approximately 3 percent of IP Phones are involved in a blind transfer on a system at any given time. Therefore, IP Tone uses a low amount of media sessions on a system.

Perform the following steps to determine the number of media sessions required to support IP Tone.

1. Determine the number of calls per hour for IP Phones (UNIStim and SIP).
   - Calls involving at least one UNIStim Phone and TPS: $C_{IPtone} = C_{2IP} + C_{1IP} + C_{STIV} + C_{STID} + C_{TSVI} + C_{TSDI} + C_{2SIPUIP}$
   - Calls involving at least one SIP Line Phone using SLG: $C_{SIPtone} = C_{2SIP} + C_{1SIP} + C_{2SIPUIP} + C_{STSV} + C_{STSD} + C_{TSVS} + C_{TSDS}$

   $Total\_IP\_calls = C_{IPtone} + C_{SIPtone}$
2. Determine the number of IP based calls that require an IP Tone.

   IP_Tone_calls = Total_IP_calls × 0.03
3. Determine the traffic load (CCS) for IP Tone. Assume the tone is required for 20 seconds.

   IPT_CCS = (IP_Tone_calls × 20) ÷ 100
4. Determine the number of IP Tone sessions required.

   Use a Poisson 1% blocking (P.01 GOS) formula or table to determine the number of IP Tone sessions you require.

Note:

IP Tone loops are the same density as TDM loops. Loop counting for determining the number of loops consumed for IP Tone is the same as TDM.

IP Conference

IP Conference provides IP based conference. The conference port calculations and loop counting remain the same as the TDM based Conference calculations. DSP resources are not required for IP Conference.

The IP Media Sessions required to support IP Conference is equal to the number of required conference ports.

Conference ports for each system = (Number of total telephones) × rcon × 0.4

IP Conference Sessions = Conference ports for each system

IP Music

You require one IP media session for every IP Music ISM ordered. You enter the number of IP Music ISM on the IP Media Services input page.

IP Recorded Announcer

You require one IP media session for every IP Recorded Announcer (IP RAN) ISM ordered. You enter the number of IP RAN ISM on the IP Media Services input page.

IP Attendant Console

You require one IP Attendant ISM for every IP Attendant console ordered. You order IP Attendant Consoles on the Consoles and terminals input page. The number of IP Attendant ISM is displayed on the IP Media Services input page.

IP Media Services sessions

The total number of IP Media Services sessions required on a system is:

MSC_Sessions = IP Conference sessions + IP Tone sessions + (3 × IP Attendant Consoles) + IP Music ISM + IP RAN ISM

Each IP Media Services maximum sessions can vary based on the hardware platform and deployment type. For example, a dedicated CP DC with 2GB of RAM can support 1000 sessions each for IP RAN, IP TONE, and IP MUSIC. A dedicated COTS2, Common Server, or CP DC with 4GB of RAM can support 4000 sessions each for IP RAN, IP TONE, and IP MUSIC.

Media Cards

Media Cards (MC32 or MC32S) do not run the Terminal Proxy Server (TPS) application. Media Cards provide DSP resources. The TPS application only runs on a Signaling Server.

All the Media Cards in a specific Media Gateway 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 42: Traffic models on page 225 summarizes the traffic models that are used in various subsystem engineering procedures.

Table 42: 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. See *Avaya Traffic Measurement Formats and Outputs Reference, NN43001-750* for a description of the TFS004 call capacity report and for information about 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

A basic call is defined as a simple, unfeatured call between two telephones. 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.

The basic call is always the call type that consumes the least CPU on the call server. For Communication Server 1000E SA / HA systems, the basic call is an i2004 to i2004 local call. For the Communication Server 1000E Co-resident Call Server and Signaling Server (Co-res CS and SS) system, and the CS 1000E TDM system, the basic call is an analog telephone to analog telephone call between two Media Gateway chassis. This call requires less CPU than a UNIStim call because the TPS application in a Co-res CS and SS system runs on the same CPU as the Call Server.

When the capacity of a switch is stated in Equivalent Basic Call (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 43: Real time capacity (EBC) by system on page 227 gives the rated capacities of the Call Server processors in systems operating CS 1000 Release 7.5.

Table 43: Real time capacity (EBC) by system

System	Capacity cph
CS 1000E (CP PIV)	740 000
CS 1000E (CP PM)	955 000
CS 1000E (CP PM Co-res CS and SS)	132 000 (see note)
CS 1000E (CP MG Co-res CS and SS)	82 000 (see note)
CS 1000E (CP DC Co-res CS and SS)	330 000 (see note)
CS 1000E (COTS2 Co-res CS and SS)	780 000 (see note)
CS 1000E (Common Server Co-res CS and SS)	780 000 (see note)

Note:

Co-res CS and SS effective throughput is significantly reduced because of high RTM factors with all calls and UCM load. The effective call throughput is the following

- CP PM - 10 000 cph
- CP DC - 15 000 cph
- CP MG - 8000 cph
- COTS2 - 20 000 cph
- Common Server - 20 000 cph

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 Enterprise Configurator 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, see System calls on page 253 and Table 53: Real time factors on page 258.

Call Server real time calculations

The system EBC divided by the processor's rated capacity (see Table 43: Real time capacity (EBC) by system on page 227) 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 on page 323.

real time algorithm

As described above, calculating the real time usage of a configuration requires information about 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. For a description of the parameters that are required and default values, see Resource calculation parameters on page 250.

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)
- SIP Line Gateway (SLG)
- Network Routing Service (NRS)
- H.323 Gatekeeper
- Network Connection Service (NCS)
- CS 1000 Element Manager Web Server
- Application Server
- Avaya Unified Communication Manager (Avaya UCM)

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 any Co-resident Signaling Server software element combination the maximum call rate supported is 10K cph.

A Signaling Server can also function as an application server for the Personal Directory, Callers List, Redial List, and Unicode Name Directory applications and Password administration. See Application server for Personal Directory, Callers List, Redial List, and Unicode Name Directory on page 244.

Table 44: Elements in Signaling Server on page 231 describes the function and engineering requirements of each element.

Table 44: 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, you can provide one extra Signaling Server 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

Element	Function and engineering requirements
	H.323 calls (see Maximum number of SIP and H.323 Virtual Trunks on page 243). Beyond that, a second Signaling Server is required. • 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.
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 3700 SIP Virtual Trunks on CP DC and COTS2, and a maximum of 1800 SIP Virtual Trunks on CP PM and COTS1 (see Maximum number of SIP and H.323 Virtual Trunks on page 243). • 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.
SIP Line Gateway (SLG)	• The SLG fully integrates Session Initiation Protocol (SIP) endpoints in the Communication Server 1000 system and extends the Communication Server 1000 features to SIP clients. • The Call Server requires Package 417 (SIPL) • The maximum SIPL users for each SLG is 3700 on CP DC, COTS2, and Common Server. The maximum SIPL users for each SLG is 1800 on CP PM and COTS1. The maximum SIPL users for a Communication Server 1000E Call Server is 11 250. In a pure IP system, the maximum SIPL users is 20 000. • You configure SIPL users as SIPL UEXT (SIPN and SIP3). SIPL users require two TNs from the Call Server, one for line TN (SIP UEXT), and one for the SIPL VTRK. There must be a 1 to 1 ratio between SIPL UEXT and SIPL VTRK TN. • SIPL redundancy can be a leader and follower configuration for a SLG node. Both Signaling Servers share the same node IP, however SIPL clients only register on the SLG node leader. The two Signaling Servers do not load share.
Network Routing Service (NRS)	• The NRS has three components: – H.323 Gatekeeper – SIP Redirect Server – SIP Proxy Server – Network Connection Service (NCS) • The NRS must reside on the Leader Signaling Server • For NRS redundancy, there are two modes: - Active-Active mode for SIP Redirect and SIP Proxy provides load balancing across two NRS servers. You must appropriately

Element	Function and engineering requirements
	engineer Active-Active mode to carry the redundant load in the case of an NRS server failure. - Primary-Secondary mode uses the Primary NRS server to handle the total call rate from all the registered endpoints. If the Primary NRS fails, the endpoints register to the Secondary NRS server to handle the calls. • The Primary and Secondary NRS Servers must be matched pairs. Unmatched vendor NRS Servers are not supported. You must use matched software configurations and engineering on each server for optimal performance. • For NRS failsafe, you must identify a Gateway Server as the NRS failsafe. • You can configure a Server as either Primary, Secondary, or Failsafe. You cannot combine multiple roles on one Server. • The NRS software limit for the total number of endpoints is 5000. An exception is SIP Proxy mode with SIP TCP transport, where the endpoints limit is 1000. • The total number of routing entries is 50 000. • The redundancy of the NRS is in a mode of 2 × N.
• 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. • The SIP Redirect Server logically routes (directly or indirectly) SIP requests to the proper destination. • The SIP Redirect Server receives requests, but does not pass the requests to another server. The SIP Redirect Server sends a response back to the SIP endpoint, indicating the IP address of the called user. The caller can directly contact the called party because the response included the address of the called user.

Element	Function and engineering requirements
• SIP Proxy Server (SPS)	• The SIP Proxy acts as both a server and a client. The SIP Proxy receives requests, determines where to send the requests, and acts as a client for the SIP endpoints to pass requests to another server.
• 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 Avaya MG 1000B, Virtual Office, and Geographic Redundancy features.
CS 1000 Element 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, Redial List, and Unicode Name Directory feature runs on the Signaling Server. • Only one database can exist in the network, and redundancy is not supported. • The database can coexist with the other software applications on a Signaling Server. However, if the number of IP users exceeds the following PD and UND limits, the database must be stored on a dedicated Signaling Server. - CP PM: PD limit of 2000 IP users, UND limit of 2000 IP users - COTS1 (HP DL320-G4 or IBM x306m): PD limit of 3000 IP users, UND limit of 3000 IP users. - COTS2 (Dell R300 or IBM x3350): PD limit of 5000 IP users, UND limit of 5000 IP users. - Common Server (HP DL360-G7): PD limit of 5000 IP users, UND limit of 5000 IP users. • The maximum number of Unicode names a dedicated server can service is 50 000. • The Application Server cannot be run on a Signaling Server at a branch office. • For more information about Personal Directory, Callers List, Redial List, and Unicode Name Directory, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-130*.
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.	

Signaling Server capacity limits

The following tables contain information about Signaling Server capacity. In these tables, the term dedicated Signaling Server means there is one Signaling Server application on a server.

The term non-dedicated Signaling Server means there is more than one Signaling Server application on a server.

Dedicated Signaling Server

Dedicated Signaling server deployments only have one application deployed (or enabled). Therefore, there is sufficient memory to support a single application with the limitations listed in the following table:

Table 45: Dedicated Signaling Server limits (one SS application per server)

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
UNIStim Phones per TPS	5000	5000	5000	5000	5000
Calls per hour (cph) per TPS	40 000	60 000	80 000	80 000	80 000
UNIStim Phone registration timing per TPS (5 min)	5000	5000	5000	5000	5000
Personal Directory users	15 000	22 500	40 000	40 000	25 000
Personal Directory cph	60 000	90 000	180 000	180 000	90 000
H323 trunks per GW	1200	1200	1200	1200	1200
H323 cph per GW	40 000	60 000	80 000	80 000	60 000
SIP trunks per SIP Signaling GW (includes ELC users)	1800	1800	3700	3700	3700
SIP cph per SIP Signaling GW	40 000	60 000	120 000	120 000	120 000
GW endpoints per NRS (UDP) - redirect or proxy	5000	5000	6000	6000	5000
GW endpoints per NRS (TCP, TCP/TLS)- redirect or proxy	1000	1000	1000	1000	1000
NRE per NRS (UDP, TCP, TCP/TLS)	50 000	50 000	300 000	300 000	50 000
H323 cph per NRS (UDP, TCP)	200 000	300 000	500 000	500 000	300 000
SIP Redirect cph per NRS (UDP)	100 000	200 000	500 000	500 000	200 000
SIP Redirect cph per NRS (TCP, TCP/TLS)	100 000	200 000	300 000	300 000	200 000
SIP Proxy cph per NRS (UDP, TCP, TCP/TLS)	50 000	100 000	200 000	200 000	100 000
OCS Clients/TR87	5000	5000	5000	5000	5000
SIPLine/SIP DECT per SLG	1800	1800	3700	3700	3700

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
SIPLine/SIP DECT cph per SLG	15 000	25 000	60 000	60 000	60 000
CS 1000 SIP Trunk Bridge cph with Media Anchoring (simultaneous conversations)	N/A	N/A	15 000	15 000	8000
SIP Trunk signaling capacity with Media Anchoring	N/A	N/A	1000	1000	1000
CS 1000 SIP Trunk Bridge cph without Media Anchoring	N/A	N/A	75 000	75 000	40 000
SIP Trunk signaling capacity without Media Anchoring	N/A	N/A	5000	5000	5000
MAS sessions G711	N/A	N/A	800 IBM 700 DELL	800	240
MAS sessions G722	N/A	N/A	445 IBM 390 DELL	445	133
MAS Sessions G729	N/A	N/A	411 IBM 360 Dell	411	123
MAS Sessions G711 + SRTP	N/A	N/A	688 IBM 602 Dell	688	206
MAS Sessions G722 + SRTP	N/A	N/A	410 IBM 360 Dell	410	123
MAS Sessions G729 + SRTP	N/A	N/A	381 IBM 334 Dell	381	114
MAS Session Call Rate (cph)	N/A	N/A	18,000	18,000	6,000
UCM number of elements	1000	1000	5000	5000	1000
Media Server Controller (MSC) cph	40 000	60 000	120 000	120 000	80 000
MSC total sessions *	1800	1800	4000	4000	4000
MSC IPConf sessions	1920	1920	1920	1920	1920
MSC IPMusic sessions	1000	1000	4000	4000	1000 if 2 GB 4000 if 4 GB
MSC IPRan sessions	1000	1000	4000	4000	1000 if 2 GB

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
					4000 if 4 GB
MSC IPTone sessions	1000	1000	4000	4000	1000 if 2 GB 4000 if 4 GB
MSC IPAttn sessions	256	256	256	256	256
* Note:	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1800	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1800	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 4000	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 4000	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 4000

Non-dedicated Signaling Server

Non-dedicated Signaling Server deployments have multiple applications deployed (or enabled) on them. The 2 GB deployments (CP PM, IBM x306m, HP DL320 G4) do not support all applications at the same time as there is a shortage of 150 MB of memory. Therefore, you must disable any combination of applications that consumes a minimum of 150 MB of memory.

In addition to the application restrictions, there are also management restrictions. Deploying a Primary UCM, Deployment Manager, EM, NRSM and Subscriber Manager on a server with only 2 GB of memory is not supported.

Note:

Two of the above management functions must be restricted.

The following is a list of recommended options for disabling applications on a 2 GB non-dedicated Signaling Server:

- Do not deploy or enable NRS or failsafe NRS and Subscriber Manager. (This option fulfills both the application and management restrictions.)
- Do not enable any MSC applications.
- Disable two Vtrk applications. For example, disable H323Gw and Presence Publisher.

  Note:

  Vtrk applications include H323Gw, SipGw, PUA (Presence Publisher), and SipLine.

	App. Size	Data	Total Memory
TPS	18 MB	85 MB	98 MB
PD	5 MB	20 MB	25 MB
Vtrk - SipGw	100 MB	30 MB	130 MB
Vtrk - H323Gw	100 MB	30 MB	130 MB
Vtrk - Presence Publisher	100 MB	30 MB	130 MB
Vtrk - Sip Line	103 MB	30 MB	133 MB
NRS (GK+SPS+NCS)	167 MB	40 MB	207 MB
Msc Announcement	33 MB	20 MB	53 MB
Msc Attendant	33 MB	5 MB	38 MB
Msc Conference	61 MB	20 MB	81 MB
Msc Music	32 MB	20 MB	52 MB
Msc Tone	32 MB	20 MB	52 MB

Figure 55: 2 GB Non-dedicated Signaling Server memory usage

Table 46: Non-dedicated Signaling Server limits (multiple SS applications per server)

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
* UNIStim Phones	1500	2000	3000	3000	2000
Personal Directory users	1500	2000	3000	3000	2000
* SIP Line Phones	800	1000	1200	1200	1000
* Virtual Trunks (H323)	800	1000	1200	1200	1000
* Virtual Trunks (SIP)	800	1000	1200	1200	1000
UCM number of elements	100	1000	2000	2000	1000
Service endpoints per NRS	100	100	100	100	100
Network Routing Entries (NRE)	1000	1000	1000	1000	1000
Media Server Controller (MSC) IPConf sessions	800	1000	1000	1000	1000
MSC IPMusic sessions	800	1000	1000	1000	1000
MSC IPRan sessions	800	1000	1000	1000	1000
MSC IPTone sessions	800	1000	1000	1000	1000
MSC IPAttn sessions	256	256	256	256	256
MSC total sessions **	800	1000	1200	1200	1000

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
Calls per hour (Sum of all applications)	15 000 sum of TPS, SipLine, Vtrk, NRS + MSC	20 000 sum of TPS, SipLine, Vtrk, NRS + MSC	30 000 sum of TPS, SipLine, Vtrk, NRS + MSC	30 000 sum of TPS, SipLine, Vtrk, NRS + MSC	20 000 sum of TPS, SipLine, Vtrk, NRS + MSC
MAS	N/A	N/A	N/A	N/A	N/A
*Note:	1500 IP users where (UNIStim + SipLine <= 1500) and (SipLine + Vtrk + ELC <= 800) and (Vtrk + MSC <= 800)	2000 IP users where (UNIStim + SipLine <= 2000) and (SipLine + Vtrk + ELC <= 1000) and (Vtrk + MSC <= 1000)	3000 IP users where (UNIStim + SipLine <= 3000) and (SipLine + Vtrk + ELC <= 2000) and (Vtrk + MSC <= 2000)	3000 IP users where (UNIStim + SipLine <= 3000) and (SipLine + Vtrk + ELC <= 2000) and (Vtrk + MSC <= 2000)	3000 IP users where (UNIStim + SipLine <= 3000) and (SipLine + Vtrk + ELC <= 1000) and (Vtrk + MSC <= 1000)
** Note:	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 800	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1000	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1200	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1200	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 1000

CS 1000E Co-resident Call Server and Signaling Server

Co-resident Signaling server deployments have the Call Server (CS) and multiple Signaling Server (SS) applications deployed (or enabled) on them. The 2 GB CP PM Co-Res CS and SS deployment does not support all applications at the same time as there is a shortage of 300 MB of memory. Therefore, you must disable any combination of applications that consumes a minimum of 300 MB of memory.

In addition to the application restrictions, there are also management restrictions. Deploying a Primary UCM, Deployment Manager, EM, NRSM and Subscriber Manager on a server with only 2 GB of memory is not supported.

Note:

Two of the above management functions must be restricted.

The following is a list of recommended options for disabling applications on a 2 GB Co-resident Call Server and Signaling Server:

- Do not enable or deploy NRS/failsafe NRS, Subscriber Manager, and one Vtrk application. (This option fulfills both the application and management restrictions.)

  Note:

  Vtrk applications include H323Gw, SipGw, PUA (Presence Publisher), and SipLine.

- Disable all MSC applications and disable one Vtrk application.
- Disable three Vtrk applications. For example, disable H323Gw, SipLine and Presence Publisher.

	App. Size	Data	Total Memory
TPS	18 MB	50 MB	68 MB
PD	5 MB	15 MB	20 MB
Vtrk - SipGw	100 MB	12 MB	112 MB
Vtrk - H323Gw	100 MB	12 MB	112 MB
Vtrk - Presence Publisher	100 MB	12 MB	112 MB
Vtrk - Sip Line	103 MB	12 MB	115 MB
NRS (GK+SPS+NCS)	167 MB	40 MB	207 MB
Msc Announcement	33 MB	10 MB	43 MB
Msc Attendant	33 MB	5 MB	38 MB
Msc Conference	61 MB	10 MB	71 MB
Msc Music	32 MB	10 MB	42 MB
Msc Tone	32 MB	10 MB	42 MB

Figure 56: 2 GB Co–resident Call Server and Signaling Server memory usage

Table 47: CS 1000E Co-resident Call Server and Signaling Server limits

	CP MG 128	CP PM	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360 G7)	CP DC
ACD Agents (IP agents, IP trunks)	200	200	200	200	200
* UNIStim Phones	700	1000	1000	1000	1000
Personal Directory users	700	1000	1000	1000	1000
* SIP Line Phones	400	400	1000	1000	400

	CP MG 128	CP PM	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360 G7)	CP DC
* Virtual Trunks (H323 and/or SIP)	400	400	400	400	400
TDM	800	128 Branch Office 800 stand alone	128 Branch Office 800 stand alone	128 Branch Office 800 stand alone	128 Branch Office 800 stand alone
PRI Spans	16	16	16	16	16
UCM Elements	100	100	100	100	100
Media Gateways (IPMG)	5	5	5	5	5
Service/GW endpoints on NRS	5	5	5	5	5
NRE on NRS	20	20	20	20	20
OCS TR87	700	1000	1000	1000	1000
Media Server Controller (MSC) IPConf sessions	400	400	400	400	400
MSC IPMusic sessions	400	400	400	400	400
MSC IPRan sessions	400	400	400	400	400
MSC IPTone sessions	400	400	400	400	400
MSC IPAttn sessions	256	256	256	256	256
MSC total sessions **	400	400	400	400	400
Calls per hour	8 000 sum of CS + NRS + MSC	10 000 sum of CS + NRS + MSC	20 000 sum of CS + NRS + MSC	20 000 sum of CS + NRS + MSC	15 000 sum of CS + NRS + MSC
* Note:	(UNIStim + SipN + Sip3 <= 700 sets and TDM <= 800 and (MSC + VTRK + ELC <=400)	(UNIStim + SipN + Sip3 <= 1000 sets AND TDM <= 800 and (MSC + VTRK + ELC <=400)	(UNIStim + SipN + Sip3 <= 1000 sets AND TDM <= 800 and (MSC + VTRK + ELC <=400)	(UNIStim + SipN + Sip3 <= 1000 sets AND TDM <= 800 and (MSC + VTRK + ELC <=400)	(UNIStim + SipN + Sip3 <= 1000 sets AND TDM <= 800 and (MSC + VTRK + ELC <=400)

	CP MG 128	CP PM	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360 G7)	CP DC
** Note:	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 400	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 400	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 400	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 400	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 400

MG 1000B Co-resident Call Server and Signaling Server

Table 48: MG 1000B Co-resident Call Server and Signaling Server limits

	CP MG 32	CP MG 128	CP PM CoRes	CP DC CoRes
ACD Agents (IP agents, IP trunks)	20	200	200	200
UniStim sets *	100	400	400	400
PD users	100	400	400	400
Sip Line sets *	0	0	0	0
vtrks (H323 and/or SIP) *	400	400	400	400
TDM	32	128	128	128
PRI Spans	1	16	16	16
UCM Elements	N/A	N/A	N/A	N/A
Media Gateways (IPMG) **	1	5	5	5
HD PRI GW	0	2	2	2
NRS	N/A	N/A	N/A	N/A
Media Server Controller IPConf sessions	10	20	20	20
MSC IPMusic sessions	30	120	120	120
MSC IPRan sessions	30	120	120	120
MSC IPTone sessions	10	20	20	20
MSC IPAttn sessions	16	64	64	64
MSC total sessions ***	100	350	350	350

	CP MG 32	CP MG 128	CP PM CoRes	CP DC CoRes
Calls per hour	8,000 sum of CS + MSC	8,000 sum of CS + MSC	10,000 sum of CS + MSC	15,000 sum of CS + MSC
* Note:	(UniStim + SipLine <= 100 sets and TDM <= 32 and (MSC + VTRK + ELC <=400)	(UniStim + SipLine <= 400 sets and TDM <= 128 and (MSC + VTRK + ELC <=400)	(UniStim + SipLine <= 400 sets and TDM <= 128 and (MSC + VTRK + ELC <=400)	(UniStim + SipLine <= 400 sets and TDM <= 128 and (MSC + VTRK + ELC <=400)
** Note:	single 4 slot chassis	Media Gateway + HD GW <= 5	Media Gateway + HD GW <= 5	Media Gateway + HD GW <= 5
*** Note:	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 100	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 350	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 350	MSC = (IPConf + IPRan + IPTone + IPMusic + IPAttn) <= 350

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 for CP PM or COTS1, and 3700 for CP DC, COTS2, or Common Server.

- 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 CP PM or COTS1 Signaling Server can support up to 1800 Virtual Trunks. Each CP DC, COTS2, or Common Server Signaling Server can support up to 3700 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 49: Maximum number of Virtual Trunks, per Signaling Server on page 244 gives examples of the maximum number of Virtual Trunks supported.

Table 49: Maximum number of Virtual Trunks, per Signaling Server

SIP	H.323*			Total Virtual Trunks
	Incoming	Outgoing	Total H.323	
3700	0	0	0	3700
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 = 3700

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, Redial List, and Unicode Name Directory

There is one Application server for the Personal Directory, Callers List, Redial List, and Unicode Name Directory features within a Communication Server 1000 system. These applications cannot be divided to run on separate Application servers.

- 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 Unicode Name Directory feature is available on the Application server. The Personal Directory, Callers List, and Redial List can operate without the Unicode Name Directory. However, to operate the Unicode Name Directory you must configure Personal Directory, Callers List, and Redial List. A dedicated PD/UND server can support 50 000 UND users.

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. The

Personal Directory, Callers List, and Redial List database can co-reside with other applications (TPS, H.323/SIP Gateways, Element Manager). For more information about Co-resident limits, see Table 59: Signaling Server algorithm constants on page 272. 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.

There is no redundancy for the Signaling Server dedicated to the Personal Directory, Callers List, Redial List, and Unicode Name Directory. If that Application Signaling Server fails, the system loses those applications.

Software configuration capacities

The tables in Design parameters on page 197 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 50: CS 1000E capacities summary on page 246 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 50: CS 1000E capacities summary

System	Total number of telephones			
	Pure TDM (no trunking)	**Pure IP (UNIStim) Access to PSTN**	**Pure IP (SIPLine) Access to PSTN**	**Mixed IP and TDM Access to PSTN**
CS 1000E (CP PIV) SA/HA	8000	22 500	2x(SipN+Sip3) <= 22 500	8000* TDM (UNIStim + 2x(SipN+Sip3) <= 22 500)*
CS 1000E (CP PM) SA/HA	8000	22 500	2x(SipN+Sip3) <= 22 500	8000* TDM (UNIStim + 2x(SipN+Sip3) <= 22 500)*
CS 1000E (CP PM) SA/HA (IP only)	N/A	40 000 (PSTN through Vtrk)	2x(SipN+Sip3) <= 40 000	N/A
CS 1000E TDM (CP PM, CP MG 128)	800	N/A	N/A	N/A
CS 1000E (Co-res CS and SS) (CP PM, COTS2, Common Server, CP DC)	800	1000	(SipN+Sip3) <= 1000	800* TDM (UNIStim + (SipN +Sip3 <= 1000)**
CS 1000E (Co-res CS and SS) (CP MG 128)	800	700	(SipN+Sip3) <= 700	800* TDM (UNIStim + (SipN +Sip3 <= 700)**

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. Enterprise Configurator and publications are used to calculate practical values preconfiguration. Values beyond these limits must be engineered.
TPS requires Signaling Servers.
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.
** The CS 1000E Co-res CS and SS system (SipN+Sip3) must always be <=1000. For example, a Co-res CS and SS system can have 600 UNIStim telephones and 400 SIP Line telephones.

Zone/IP Telephony Node Engineering

For information about Zone/IP Telephony Node Engineering, see *Avaya Communication Server 1000M and Meridian 1 Large System Planning and Engineering, NN43021-220*.

Chapter 14: Resource calculations

Contents

This chapter contains the following topics:

Introduction

This chapter describes the algorithms implemented by the Enterprise Configurator 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 on page 207 and Application engineering on page 323.

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 51: Resource calculation parameters on page 250 describes the parameters you use in resource calculations.

Table 51: Resource calculation parameters

Parameter	Description	Default value
$Telephone_{CCS}$	Centi-Call Seconds (CCS) for each standard telephone	5 CCS
$NBTelephone_{CCS}$	CCS for each nonblocking telephone	18 CCS
ACD_{CCS}	CCS for each Automatic Call Distribution (ACD) telephone	33 CCS
TRK_{CCS}	CCS for each Time-Division Multiplexing (TDM) trunk	28 CCS
$VTRK_{CCS}$	CCS for each Virtual trunk	28 CCS
$CP1_{CSS}$	CCS for each local CallPilot port	26 CCS
$CP2_{CCS}$	CCS for each network CallPilot port	26 CCS
$APPL_{CCS}$	CCS for each application port	26 CCS
Intraoffice ratio (R_I)	The portion of telephone to telephone calls, from the total number of calls.	0.30
Tandem ratio (R_T)	The portion of trunk to trunk calls, from the total number of calls.	0.05
Incoming ratio (I)	The portion of trunk to telephone calls, from the total number of calls.	0.40
Outgoing ratio (O)	The portion of telephone to trunk calls, from the total number of calls $O = 1 - R_I - R_T - I$ The sum of all four traffic ratios (R_I, R_T, I, O) must equal 1	0.25 calculated on the values for R_I, R_T, and I
AHT_{SS}	Average holding time (AHT) for telephone-to-telephone calls	60 seconds
AHT_{TS}	AHT for trunk to telephone calls	150 seconds
AHT_{ST}	AHT for telephone to trunk calls	150 seconds
AHT_{TT}	AHT for trunk to trunk calls	180 seconds
AHT_{AGT}	AHT for ACD agent calls	180 seconds
AHT_{CP}	AHT for CallPilot calls	40 seconds

Parameter	Description	Default value
AHT_{MICB}	AHT for Integrated Conference Bridge calls	1800 seconds
AHT_{MIRAN}	AHT for Integrated Recorded Announcement calls	90 seconds
AHT_{MIPCD}	AHT for Integrated Call Director calls	60 seconds
AHT_{MICA}	AHT for Integrated Call Announcement calls	180 seconds
AHT_{MIVS}	AHT for Integrated Voice Services calls	90 seconds
r_{con}	Conference loop ratio (number of conference loops / total number of loops)	0.07

Resource calculation equations

Table 52: Resource calculation equations on page 251 describes the equations you use in resource calculations. Evaluate the equations in the order shown, as other calculations require the results of the previous calculations.

Table 52: Resource calculation equations

Name	Description	Equation
L_{TDM}	Total TDM telephone CCS, including analog, digital and line-side T1/E1 ports	L_{TDM} = ((number of digital telephones + number of analog telephones + number of line-side T1/E1 ports) × $Telephone_{CCS}$) + (number of nonblocking telephones × $NBTelephone_{CCS}$)
L_{IP}	Total UNIStim IP Phone CCS, including wireless IP telephones	L_{IP} = (number of UNTIStim IP Phones - number of IP ACD agents) × $Telephone_{CCS}$
L_{ACD}	Total TDM ACD agent CCS	L_{ACD} = (number of TDM ACD agents) × ACD_{CCS}
L_{ACDIP}	Total IP ACD agent CCS	L_{ACDIP} = (number of UNIStim ACD agents) × ACD_{CCS}
L_{DECT}	Total Digital Enhanced Cordless Telecommunications (DECT) telephone CCS, excluding SIP-DECT	L_{DECT} = (number of DECT telephones) × $Telephone_{CCS}$
L_{IPW}	Total IP Wireless 802.11 telephone CCS	L_{IPW} = (number of Wireless 802.11 telephones) × $Telephone_{CCS}$

Name	Description	Equation
L_{SIPL}	Total SIP Line telephone CCS, including SIP-DECT	L_{SIPL} = (number of SIPN telephones + number of SIP3 telephones) × $Telephone_{CCS}$
ACD_{adj}	ACD CCS adjustment for TDM agents (see paragraph below table)	ACD_{adj} = (number of TDM ACD agents × $NBTelephone_{CCS}$)
L_{CCS}	Total line CCS. The sum of all telephone CCS	$L_{CCS} = (L_{TDM} + L_{IP} + L_{ACD} + L_{ACDIP} + L_{DECT} + L_{IPW} + L_{SIPL}) - ACD_{adj}$
T_{TDM}	Total TDM trunk CCS, including analog and digital trunks	T_{TDM} = (number of analog trunks + number of digital trunks) × TRK_{CCS}
HVT_{CCS}	Total H323 trunk CCS	HVT_{CCS} = (number of H323 trunks) × TRK_{CCS}
SVT_{CCS}	Total SIP trunk CCS	SVT_{CCS} = (number of SIP trunks) × TRK_{CCS}
VT_{CCS}	Total Virtual Trunk CCS	$VT_{CCS} = HVT_{CCS} + SVT_{CCS}$
T_{TCCS}	Total trunk CCS	$T_{TCCS} = T_{TDM} + VT_{CCS}$
V_H	Percentage of H323 trunk CCS from the total Virtual Trunk CCS	$V_H = HVT_{CCS} \div VT_{CCS}$
V_S	Percentage of SIP trunk CCS from the total Virtual Trunk CCS	$V_S = SVT_{CCS} \div VT_{CCS}$
V	Percentage of Virtual Trunk CCS from the total trunk CCS	$V = VT_{CCS} \div T_{TCCS}$
P_U	Ratio of UNIStim telephone CCS to total telephone CCS	$P_U = (L_{IP} + L_{ACDIP}) \div L_{CCS}$
P_S	Ratio of SIP Line telephone CCS to total telephone CCS	$P_S = L_{SIPL} \div L_{CCS}$
P_{IP}	Ratio of IP telephone CCS to total telephone CCS	$P_{IP} = (L_{IP} + L_{ACDIP} + L_{SIPL}) \div L_{CCS} = P_U + P_S$
WAHT	Weighted average holding time (WAHT) in seconds	$WAHT = (R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$
Total_telephones	Total number of telephones on the system	Total_telephones = number of TDM telephones + number of UNIStim telephones + number of SIPN telephones + number of SIP3 telephones + number of MDECT telelphones + number of SIP-DECT telephones + number of BRI telephones + number of MobileX users + number of MC3100 users

Name	Description	Equation
r_{DTP}	Ratio of converged desktop telephones to total number of telephones	r_{DTP} = number of telephones with converged desktop ÷ Total_telephones
V_{DCCS}	Converged desktop CCS	$V_{DCCS} = L_{CCS} \times r_{DTP}$
MOP	Percentage of Microsoft Converged Office users	MOP = number of telephones with Microsoft Converged Office ÷ Total_telephones
IPSECP	IP security usage	If any IP security features are enabled IPSECP = 1, otherwise IPSECP = 0
CP1	Local CallPilot CCS	CP1 = number of local CallPilot ports × $CP1_{CCS}$
CP2	Network CallPilot CCS	CP2 = number of network CallPilot ports × $CP2_{CCS}$
MC3100_P	Percentage of MC3100 users	MC3100_P = MC3100_users ÷ Total_telephones
MobileX_P	Percentage of MobileX users	MobileX_P = MobileX_users ÷ Total_telephones

The ACD traffic for TDM terminations is integral in L_{TDM} for all systems. Large systems can contain both standard and nonblocking telephones. You must enter ACD agents in the nonblocking telephone count (for line card provisioning), therefore adjust the CSS using the nonblocking CCS rate.

L_{IP} is correct for IP ACD agents. L_{IP} does not require a CSS adjustment because all IP Phones are nonblocking.

Total system traffic

System traffic is the sum of traffic from the sources. The total system CCS is the sum of telephone and trunk CCS.

$T_{CCS} = L_{CCS} + T_{TCCS}$

System calls

The total number of calls the system must be engineered to handle is given by:

$T_{CALL} = 0.5 \times T_{CCS} \times 100 \div WAHT$

Traffic equations and penetration factors

Total system calls comprise four different types of traffic:

1. Telephone-to-telephone (Intraoffice calls (C_{SS}) on page 254)
2. Trunk-to-trunk (Tandem calls (C_{TT}) on page 255)
3. Telephone-to-trunk (Originating/outgoing calls (C_{ST}) on page 255)
4. Trunk-to-telephone (Terminating/incoming calls (C_{TS}) on page 256)

1. Intraoffice calls (C_{SS})

   C_{SS} = Total calls (T_{CALL}) × Intraoffice ratio (R_I)

   This parcel can be further broken down to six types:

   a. Intraoffice UNIStim IP to UNIStim IP calls (UIPtoUIP)

      $C_{2IP} = C_{SS} \times P_U \times P_U$ (require no DSP, no VT)

      P_UIPtoUIP = $C_{2IP} \div T_{CALL}$ P_UIPtoUIP is the penetration factor for the intraoffice UNIStim IP to UNIStim IP calls

   b. Intraoffice UNIStim IP to TDM telephone calls (UIPtoL)

      $C_{1IP} = C_{SS} \times 2 \times P_U \times (1 - P_{IP})$ (require DSP, no VT)

      P_UIPtoL = $C_{1IP} \div T_{CALL}$ P_UIPtoL is the penetration factor for the intraoffice UNIStim IP to TDM telephone calls

   c. Intraoffice TDM telephone to TDM telephone calls (LtoL)

      $C_{NoIP} = C_{SS} \times (1 - P_{IP})^2$ (require no DSP, no VT)

      P_LtoL = $C_{NoIP} \div T_{CALL}$ P_LtoL is the penetration factor for the intraoffice TDM to TDM calls

   d. Intraoffice SIP Line to SIP Line calls (SIPtoSIP)

      $C_{2sip} = C_{SS} \times P_S^2$ (require no DSP, no VT)

      P_SIPtoSIP = $C_{2sip} \div T_{CALL}$ P_SIPtoSIP is the penetration factor for the intraoffice SIP Line to SIP Line calls

   e. Intraoffice SIP Line to UNIStim IP calls (SIPtoUIP)

      $C_{2sipuip} = C_{SS} \times 2 \times P_S \times P_U$ (require no DSP, no VT)

      P_SIPtoUIP = $C_{2sipuip} \div T_{CALL}$ P_SIPtoUIP is the penetration factor for the intraoffice SIP Line to UNIStim IP calls

   f. Intraoffice SIP Line to TDM calls (SIPtoL)

      $C_{1sip} = C_{SS} \times 2 \times P_S \times (1 - P_{IP})$ (require DSP, no VT)

P_SIPtoL = $C_{1sip} \div T_{CALL}$ P_SIPtoL is the penetration factor for the intraoffice SIP Line to TDM calls

2. Tandem calls (C_{TT})

   C_{TT} = Total calls × Tandem ratio = $T_{CALL} \times R_T$

   The tandem calls can be further broken down into:

   a. Tandem VT to TDM trunk calls (VTtoTr)

      $C_{T1VT} = 2 \times C_{TT} \times V \times (1 - V)$ (require DSP and VT)

      P_VTtoTr = $C_{T1VT} \div T_{CALL}$ P_VTtoTr is the penetration factor for the tandem VT to TDM trunk calls

   b. Tandem TDM trunk to TDM trunk calls (TrtoTr)

      $C_{T2NoVT} = C_{TT} \times (1 - V)^2$ (require no DSP, no VT)

      P_TrtoTr = $C_{T2NoVT} \div T_{CALL}$ P_TrtoTr is the penetration factor for the tandem TDM trunk to TDM trunk calls

   c. Tandem VT (H.323) to VT (SIP) calls (VhtoVs)

      C_{T2HS}= CTT × $V^2 \times V_H \times V_S \times 2 \times 2$ (require VT)

      where V_H is the fraction of H.323 trunks to total VTs, and V_S 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, P_VhtoVs = 0.

      P_VhtoVs = $C_{T2HS} \div T_{CALL}$ P_VhtoVs is the penetration factor for the tandem VT (H.323) to VT (SIP) calls

3. Originating/outgoing calls (C_{ST})

   C_{ST} = Total calls × Outgoing ratio = $T_{CALL} \times O$

   Originating/outgoing calls can be further broken down into:

   a. UNIStim IP to VT calls (UIPtoVT)

      $C_{STIV} = C_{ST} \times P_U \times V$ (require no DSP, no VT)

      P_UIPtoVT = $C_{STIV} \div T_{CALL}$ P_UIPtoVT is the penetration factor for the outgoing UNIStim IP to VT calls

   b. UNIStim IP to TDM trunk calls (UIPtoTr)

      $C_{STID} = C_{ST} \times P_U \times (1 - V)$ (require DSP, no VT)

      P_UIPtoTr = $C_{STID} \div T_{CALL}$ P_UIPtoTr is the penetration factor for the outgoing UNIStim IP to TDM trunk calls

   c. TDM telephone to VT calls (LtoVT)

      $C_{STDV} = C_{ST} \times (1 - P_{IP}) \times V$ (require DSP, VT)

$P\_LtoVT = C_{STDV} \div T_{CALL}$ P_LtoVT is the penetration factor for the outgoing TDM telephone to VT calls

d. TDM telephone to TDM trunk calls (LtoTr) $C_{STDD} = C_{ST} \times (1 - P_{IP}) \times (1 - V)$ (require no DSP, no VT)

   $P\_LtoTr = C_{STDD} \div T_{CALL}$ P_LtoTr is the penetration factor for the outgoing TDM telephone to TDM trunk calls

e. SIP Line to VT calls (SIPtoVT) $C_{STSV} = C_{ST} \times P_S \times V$ (require no DSP, VT)

   $P\_SIPtoVT = C_{STSV} \div T_{CALL}$ P_SIPtoVT is the penetration factor for the outgoing SIP Line to VT calls

f. SIP Line to TDM trunk calls (SIPtoTr) $C_{STSD} = C_{ST} \times P_S \times (1 - V)$ (require DSP, no VT)

   $P\_SIPtoTr = C_{STSD} \div T_{CALL}$ P_SIPtoTr is the penetration factor for the outgoing SIP Line to TDM trunk calls

4. Terminating/incoming calls (C_{TS})

   C_{TS} = Total calls × Incoming ratio = $T_{CALL} \times I$

   Terminating/incoming calls can be further broken down into:

   a. VT to TDM telephone calls (VTtoL)

      $C_{TSVD} = C_{TS} \times V \times (1 - P_{IP})$ (require DSP, VT)

      $P\_VTtoL = C_{TSVD} \div T_{CALL}$ P_VTtoL is the penetration factor for the incoming VT to TDM telephone calls

   b. VT to UNIStim IP telephone calls (VTtoUIP)

      $C_{TSVI} = C_{TS} \times V \times (P_U)$ (require no DSP, VT)

      $P\_VTtoUIP = C_{TSVI} \div T_{CALL}$ P_VTtoUIP is the penetration factor for the incoming VT to UNIStim IP telephone calls

   c. TDM trunk to UNIStim IP telephone calls (TrtoUIP)

      $C_{TSDI} = C_{TS} \times (1 - V) \times (P_U)$ (require DSP, no VT)

      $P\_TrtoUIP = C_{TSDI} \div T_{CALL}$ P_TrtoUIP is the penetration factor for the incoming TDM trunk to UNIStim IP telephone calls

   d. TDM trunk to TDM telephone calls (TrtoL)

      $C_{TSDD} = C_{TS} \times (1 - V) \times (1 - P_{IP})$ (require no DSP, no VT)

      $P\_TrtoL = C_{TSDD} \div T_{CALL}$ P_TrtoL is the penetration factor for the incoming TDM trunk to TDM telephone calls

   e. VT to SIP Line calls (VTtoSIP)

      $C_{TSVS} = C_{TS} \times V \times P_S$ (require DSP, no VT)

$P\_VTtoSIP = C_{TSVS} \div T_{CALL}$ P_VTtoSIP is the penetration factor for the incoming VT to SIP Line calls

f. TDM trunk to SIP Line calls (TrtoSIP)

$C_{TSDS} = C_{TS} \times (1 - V) \times P_S$ (require no DSP, no VT)

$P\_TrtoSIP = C_{TSDS} \div T_{CALL}$ P_TrtoSIP is the penetration factor for the incoming TDM trunk to SIP Line calls

Resource use equations

The following equations, summing different types of traffic, are used to calculate the required TPS and SLG resources.

- Calls involving at least one UNIStim IP Phone and therefore using TPS:

  $C_{IP} = (2 \times C_{2IP}) + C_{1IP} + C_{STIV} + C_{STID} + C_{TSVI} + C_{TSDI} + C_{2SIPUIP}$
- Calls that require at least one SIP Line Phone and therefor using SLG:

  $C_{SIP} = (2 \times C_{2SIP}) + C_{1SIP} + C_{2SIPUIP} + C_{STSV} + C_{STSD} + C_{TSVS} + C_{TSDS}$

Real time calculations

This section describes the following real time calculations:

- System EBC without applications on page 260
- Feature and applications EBCs on page 260
- Call Server utilization on page 262
- CPU real time conversion for upgrades on page 262

The real time required to process a basic telephone to basic telephone call is an Equivalent Basic Call (EBC), which is the unit used to measure other, more complicated feature calls. The basic call type varies for each system, and is based on the call requiring the least amount of Call Server CPU. 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 21 major combinations of telephone and trunk types of calls in the system. The real time factor of each call type is denoted by f_i (i = 1 to 21). In addition, there are standard real time factors for applications and features. Table 53: Real time factors on page 258 provides the real time factors.

Table 53: Real time factors

Type of call	Real time factor	Avaya CS 1000E SA/HA	Avaya CS 1000E Co-res + SLG
Intraoffice calls:			
UNIStim IP telephone to UNIStim IP telephone	(f_1)	0.0	4.90
UNIStim IP telephone to TDM telephone	(f_2)	1.41	1.86
TDM telephone to TDM telephone	(f_3)	0.81	0.03
Tandem calls:			
Virtual Trunk to TDM trunk	(f_4)	1.14	3.65
TDM trunk to TDM trunk	(f_5)	1.20	2.85
H.323 Virtual Trunk to SIP Virtual Trunk	(f_6)	1.09	4.45
Originating/outgoing calls:			
UNIStim IP telephone to Virtual Trunk	(f_7)	1.20	5.18
UNIStim IP telephone to TDM trunk	(f_8)	1.16	4.96
TDM telephone to Virtual Trunk	(f_9)	2.44	3.95
TDM telephone to TDM trunk	(f_{10})	1.25	4.36
Terminating/incoming calls:			
Virtual Trunk to TDM telephone	(f_{11})	1.72	2.86
Virtual Trunk to IP telephone	(f_{12})	0.97	3.72
TDM trunk to UNIStim IP telephone	(f_{13})	1.25	0.74
TDM trunk to TDM telephone	(f^{14})	1.34	0.06
SIPL to SIP telephone	(f^{15})	2.72	4.80
SIPL to UNIStim IP telephone	(f^{16})	1.36	2.04
SIPL to TDM telephone	(f^{17})	1.78	2.50
SIPL to Virtual Trunk	(f^{18})	1.97	5.60
SIPL to TDM Trunk	(f^{19})	2.25	2.14
Virtual Trunk to SIP telephone	(f^{20})	2.17	5.97
TDM Trunk to SIPL	(f^{21})	3.57	4.97
Application/feature calls:			
ACD agent without Symposium	(f_{ACD})	0.13	0.13

Type of call	Real time factor	Avaya CS 1000E SA/HA	Avaya CS 1000E Co-res + SLG
Symposium	(f_{SYM})	5.70	5.70
CallPilot	(f_{CP})	1.70	1.70
Avaya Integrated Conference Bridge	(f_{MICB})	1.59	1.59
Avaya Integrated Recorded Announcer	(f_{MIRAN})	0.63	0.63
Avaya Integrated Call Assistant	(f_{MICA})	0.57	0.57
Avaya Hospitality Integrated Voice Service	(f_{MIVS})	0.57	0.57
Avaya Integrated Call Director	(f_{MIPCD})	0.63	0.63
IP Recorded Announcer on Media Server	(f_{IPRAN_MS})	1.2	1.2
Media Services based tones	(f_{IPTDS})	0.8	0.8
BRI ports	(f_{BRI})	0.12	0.12
MDECT telephone	(f_{DECT})	4.25	4.25
Intraoffice CDR	(f_{ICDR})	0.44	0.44
Incoming CDR	(f_{CCDR})	0.32	0.32
Outgoing CDR	(f_{OCDR})	0.32	0.32
Tandem CDR	(f_{TAN})	0.44	0.44
CPND factor	(f_{CPND})	0.20	0.20
Converged Desktop factor	(f_{DTP})	2.33	2.33
Error term – minor feature overhead	(f_{OVRH})	0.25	0.25
Microsoft Converged Office factor	(f_{MO})	2.33	2.33
IP Security factor	(f_{IPSEC})	0.33	0.33
MC3100 factor	(f_{MC3100})	4.78	4.78
MobileX factor	($f_{mobileX}$)	0.67	0.67
Extend Local Calls factor	(f_{ELC})	4.36	4.36

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 with which 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. See Traffic equations and penetration factors on page 254 for the equations to calculate penetration factors for the 21 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:

RTM = 1 + Error_term + (P_UIPtoUIP × f_1) + (P_UIPtoL × f_2) + (P_LtoL × f_3) + (P_VTtoTr × f_4) + (P_TrtoTr × f_5) + (P_VhtoVs × f_6) + (P_UIPtoVT × f_7) + (P_UIPtoTr × f_8) + (P_LtoVT × f_9) + (P_LtoTr × f_{10}) + (P_VTtoL × f_{11}) + (P_VTtoUIP × f_{12}) + (P_TrtoUIP × f_{13}) + (P_TrtoL × f_{14}) + (P_SIPtoSIP × f_{15}) + (P_SIPtoUIP × f_{16}) + (P_SIPtoL × f_{17}) + (P_SIPtoVT × f_{18}) + (P_SIPtoTr × f_{19}) + (P_VTtoSIP × f_{20}) + (P_TrtoSIP × f_{21})

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.

System EBC without applications

System EBC = (Total system calls × real time Multiplier) SEBC = (T_{CALL} × RTM)

Feature and applications EBCs

Table 54: Feature and applications EBC on page 260 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 54: Feature and applications EBC

EBC feature or application	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) × ACD_{CCS} × 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) × ACD_{CCS} × 100 ÷ AHT_{AGT}
CallPilot	(CP1 + CP2) × 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}

EBC feature or application	Calculation
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}
IP_RAN	(((IP RAN ports + IP MUS ports) × 26) × 100) ÷ 90) × f_{IPRAN}
IP_TDS	IP Tone Calls × f_{MSTDS}
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 × f_{MO}
IP Security (IPSEC)	(C_{ss} + C_{TT} + C_{TS}) × P × ipsecp × f_{IPSEC}
MC3100	(C_{ss} + C_{TT} + C_{ST} + C_{TS}) × MC3100_P × f_{MC3100} where MC3100_P = MC3100_users / (MC3100_users + total number of set and extensions)
MobileX	(C_{ss} + C_{TT} + C_{ST} + C_{TS}) × MobileX_P × $f_{MobileX}$ where MobileX_P = MobileX_users / (total number of set and extensions)
Extend Local Calls (ELC)	(C_{ss} × ELC_P × f_{ELC} where ELC_P = ELC_users / (total number of telephones and extensions)

Feature and application EBC defines as: FAEBC = ACD_EBC + Symposium_EBC + CallPilot_EBC + InternalCDR_EBC + IncomingCDR_EBC + OutgoingCDR_EBC + TandemCDR_EBC + MICB_EBC + MIRAN_EBC + MIPCD_EBC + MICA_EBC + MIVS_EBC + BRI_EBC + MDECT_EBC + CPND_EBC + CD_EBC + MO_EBC + IPSEC_EBC + MC3100_EBC + MobileX_EBC + IP_RAN_EBC + IP_TDS_EBC + ELC_EBC

Call Server utilization

Real Time Usage (RTU) is expressed as a percentage of the rated EBC capacity of the CPU. RTU = (SEBC + FAEBC) ÷ Rated_EBC × 100 Where Rated_EBC value is from Table 43: Real time capacity (EBC) by system on page 227.

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 will be 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

- 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, set 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 55: Software release degradation factors (SWRC) on page 263 lists the software release degradation factors for supported software upgrades.

Table 55: Software release degradation factors (SWRC)

From	Degradation factor (%)
	to CS 1000E Release 7.5
Release 18	310
Release 19	291
Release 20B	220
Release 21B	190
Release 22	149
Release 23	130
Release 23C	125
Release 24B	81
Release 25B	69
Succession Release 2	66
Succession Release 3	62
CS 1000 Release 4	55
CS 1000 Release 4.5	42
CS 1000 Release 5	23
CS 1000 Release 5.5	17
CS 1000 Release 6.0	11
CS 1000 Release 7.0	6

Table 56: Ratio of existing processor capacity to new processor capacity (CPTU) on page 263 gives capacity ratio values for supported processor upgrades.

Table 56: Ratio of existing processor capacity to new processor capacity (CPTU)

From CPU type	EBC Ratio		
	To CP PII	To CP PIV	To CP PM
CP PII	1.00	0.32	0.30
CP PIV	–	1.00	0.667
CP PM	–	–	1.00

DSP/Media Card calculations

DSP resources are provided by Media Cards and MGC DSP daughterboards. The total DSP requirement is the sum of DSP requirements for various functions, which are calculated separately for each Media Gateway.

- DSP ports for Conference on page 265
- DSP ports for general traffic on page 266
- DSP ports for major applications on page 266
- Special treatment for nonblocking access to DSP ports on page 267
- Total Media Gateway DSP requirements on page 267
- Simplified Media Gateway DSP calculations on page 268

For reasons explained in the Traffic models on page 225section, the Erlang B model is used to calculate DSP port requirements.

The DSP port requirement must be calculated in increments of 32. The Gateway Controller, MC32, and MC32S provide DSPs in groups of 32. DSP calculations that refer to a Media Card equal a bank of 32 DSP ports. The 32 DSP ports can be provided by a MC32, MC32S, or DSP resources from a Gateway Controller. For example, an MGC card with 128 DSP ports is the equivalent of 4 Media Cards.

Table 57: Erlang B and Poisson values, in 32-port increments on page 264 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 on page 269).

Table 57: 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 Media Gateways, 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, 2000 CCS requires 96 DSP ports to provide a P.01 GoS (2000 >1822) as calculated from Table 57: Erlang B and Poisson values, in 32-port increments on page 264. In this example, you must provide 3 Media Cards, or a 96 DSP ports.

For information about allocating Media Cards to Media Gateways, see Assigning loops and card slots in the Communication Server 1000E on page 379.

DSP ports for Conference

A DSP channel is required for each telephone joining a conference call. The more telephones in the system, the higher the demand for DSP channels to access the conference feature.

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:

Number of conference ports = [(Total number of telephones) × r_{Con} × 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.

Number of conference loops = ROUNDUP [Number of Conference ports ÷ 30]

Number of DSP ports for Conference = Number of Conference loops × 32 (You require 1 Media Card or 32 DSP ports for each defined conference loop).

Note:

For IP Media Services, no dedicated DSP resources are required to support IP conference.

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 a Media Gateway, the DSPs required to support that digital trunk card must be placed into the same Media Gateway.

Number of DSP ports for digital trunks = Number of digital trunk ports

DSP ports for general traffic

There are two steps to calculate the number of DSP ports required for general traffic in each Media Gateway:

1. Calculate the number of CCS for all standard cards in the Media Gateway.

   Media Gateway standard CCS = (Number of standard telephones × CCS for each telephone) + (Number of standard analog trunks × CCS for each trunk) + (Number of standard consoles × 100 CCS).

   A standard console requires 4 DSP ports that accounts for approximately 100 CCS of usage.
2. Using the Erlang B table for P.01 GoS (see Table 57: Erlang B and Poisson values, in 32-port increments on page 264), find the corresponding number of DSP ports required.

   Number of DSP ports for general traffic = Required number of ports for DSP CCS from Erlang B table

DSP ports for major applications

For most applications, provide one DSP port for each application port.

Table 58: DSP port requirements for applications on page 267 provides the equations to calculate the number of DSP ports required for each application. You must provide enough DSP ports in each Media Gateway that contain application cards.

Table 58: DSP port requirements for applications

Application or port type	Calculation
Integrated Recorded Announcer	Number of Integrated Recorded Announcer ports
Integrated Conference Bridge	Number of Integrated Conference Bridge ports
Integrated Call Director	Number of Integrated Call Director ports
Integrated Call Assistant	Number of Integrated Call Assistant ports
Hospitality Integrated Voice Service	Number of Hospitality Integrated Voice Service ports
BRI	Number of SILC ports = Number of BRI users × 2
CallPilot ports	(Number of local CallPilot ports) + (Number of network CallPilot ports) (see Note)
Agent Greeting ports	Number of Agent Greeting ports
Avaya CallPilot calls served by another node are treated as trunk traffic and are not included in DSP calculations for this node.	

Number of DSP ports for applications = DSP for Integrated Recorded Announcer + DSP for Integrated Conference Bridge + ... + DSP for Agent Greeting ports

Special treatment for nonblocking access to DSP ports

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 nonblocking TDM telephone requiring a DSP resource, in order to provide nonblocking access between the telephone and DSP.

The nonblocking telephones must be located in the same Media Gateway as the Media Cards providing the DSP resources for the telephones.

Number of DSP ports for nonblocking traffic = Number of nonblocking TDM telephones + number of nonblocking analog trunks + (number of nonblocking consoles × 12)

Total Media Gateway DSP requirements

Number of DSP ports for each Media Gateway = Number of DSP ports for conference equation on page 265 + Number of DSP ports for digital trunks equation on page 266 + Number of DSP ports for general traffic equation on page 266 + Number of DSP ports for applications equation on page 267 + Number of DSP ports for nonblocking traffic equation on page 267

Simplified Media Gateway DSP calculations

You must allocate DSP ports for each Media Gateway in a Communication Server 1000E system. The general DSP calculations on system wide traffic only provide an estimate of the total number of DSP ports required on a Communication Server 1000E. You can use the following rules to avoid complex DSP calculations for each Media Gateway.

Provision each Media Gateway with 128 DSP ports. This configuration allows a mix of blocking and nonblocking cards in a standard Media Gateway with certain limits. You can configure a maximum of 60 nonblocking ports in a standard Media Gateway. For example, you can configure any two of either PRI, DTI, ICB, Mgate cards, and so on. RAN/MUSIC connections also count as nonblocking ports. You can add a maximum of 30 conference ports in addition to the previously selected items.

Note:

For IP Media Services, no dedicated DSP resources are required.

Each Communication Server 1000E system can have two types of Media Gateway DSP configurations:

- maximum of 2 nonblocking resource units
- any number of blocking cards (limited by the available card slots in the Media Gateway)
- maximum of 2 standard consoles

A nonblocking resource unit can be any of the following

- 1 x PRI or digital trunk card (for example, 24-channel TMDI/T1 or 30-channel PRI/E1)
- 1 x CallPilot card
- 1 x DECT card (for example, DMC / DMC8 or DMC-E / DMC8-E)
- 1 x Agent Greeting, SECC VSC, CRQM and RAO cards
- 2 x nonblocking digital or analog line cards (includes analog card for CLASS and Reach line card)
- 2 x nonblocking analog trunk cards
- 2 x nonblocking consoles
- 1 x nonblocking line side interface card
- 30 x broadcast circuits (RAN, MUSIC)
    - Accounts for MIRAN cards. If you assign more than 60 broadcast circuits to a single MIRAN or trunk card, you must install it in a dedicated nonblocking Media Gateway. If less than 1 MIRAN or trunk card is available or assigned, you can have a maximum of 30 RAN or MUSIC for each card and install each in a standard Media Gateway.

A blocking card can be any of the following:

- CP PM and XCMC (CLASS clock)
- Standard digital or analog lines
- Standard Reach line cards
- Standard trunk cards

If there are high CCS rates for each card, you must configure the cards as nonblocking.

A nonblocking Media Gateway can also be referred to as a Dedicated Media Gateway. A nonblocking Media Gateway has one DSP port for each resource in the Media Gateway. 12 DSP ports are required for each nonblocking console.

Virtual Trunk calculations

For reasons explained in the "System capacities on page 207" chapter (see Traffic capacity engineering algorithms on page 224), the Poisson model is used to calculate trunk requirements.

Table 57: Erlang B and Poisson values, in 32-port increments on page 264 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 57: Erlang B and Poisson values, in 32-port increments on page 264) × (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}) + UIP-VT calls (C_{STIV}) + TDM telephone-VT calls (C_{STDV}) + VT-TDM telephone calls (C_{TSVD}) + VT-UIP telephone calls (C_{TSVI}) + H.323-SIP VT calls (C_{T2HS}) + SIP Line-VT calls (C_{STSV}) + VT-SIP Line calls (C_{TSVS})

   $C_{VT} = C_{T1VT} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} + C_{T2HS} + C_{STSV} + C_{TSVS}$

   Calls that require H.323 Virtual Trunks: $HC_{VT} = C_{VT} \times V_H$

   Calls that require SIP Virtual Trunks: $SC_{VT} = C_{VT} \times V_S$

   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, C_{VT} is a reduced total that excludes ACD CCS. See Special treatment for nonblocking access to DSP ports on page 267.

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 will be sufficient to handle both general and ACD traffic.

   Final Virtual Trunk CCS = (Calculated VT_{CCS} without ACD) + [(Number of IP ACD agent telephones) + (Number of TDM ACD agent telephones)] × V × ACD_{CCS} ÷ TRK_{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 will be allocated to the two groups.

   SIP Virtual Trunk CCS (SVT_{CCS}) = VT_{CCS} × v_S H.323 Virtual Trunk CCS (HVT_{CCS}) = VT_{CCS} × v_H

5. Using the Poisson table for P.01 GoS (seeTable 57: Erlang B and Poisson values, in 32-port increments on page 264 or Trunk traffic Erlang B with P.01 Grade-of-Service on page 413), 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) on page 305.

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. For information about the bandwidth table and calculating LAN/WAN bandwidth requirements, see *Avaya Converging the Data Network with VoIP Fundamentals , NN43001-260*.

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.

The Signaling Server hardware platform can be CP PM, CP DC, IBM x306m, HP DL320-G4, HP DL360-G7, IBM x3350, or Dell R300 servers. For the calculations, each variable is indexed by Signaling Server type. type index = CP PM or CP DC or COTS1 (HP DL320-G4, IBM x306m), COTS2 (IBM x3350, Dell R300), or Common Server (HP DL360-G7).

Table 59: Signaling Server algorithm constants on page 272 defines the constants you use in the Signaling Server algorithm.

Table 59: Signaling Server algorithm constants

Algorithm Constant	Description	Limit	Notes
NRC_{HL}[type_index]	NRS calls per hour	CP PM = 200 000 CP DC = 300 000 COTS1 = 300 000 COTS2= 500 000 Common Server= 500 000	Hardware limit for Signaling Server. The SIP loading factor determines SIP Redirect and SIP Proxy call rates, see Table 64: SIP mode factors on page 289.
NRC_{SL}[type_index]	NRS calls per hour	CP PM = 20 000 CP DC= 30 000 COTS1 = 30 000 COTS2 = 50 000 Common Server = 50 000	Shared limit for Signaling Server.
NRD_1[type_index]	NRS CDP + UDP entries limit	50 000 COTS2 = 300 000 Common Server = 300 000	Software limit.
NRD_{HL}[type_index]	NRS product of endpoint and CDP/UDP entries	50 000 COTS2 = 300 000 Common Server = 300 000	Hardware limit.
NRE_1[type_index]	NRS endpoints limit	5000 COTS2 = 6000 Common Server = 6000	Software limit.
SIP_Proxy_Limit[type_index]	SIP Proxy NRS endpoints limit for TCP only	1000	Software limit.
NRP_{SL}[type_index]	NRS product of endpoint and CDP/UDP entries	50 000 COTS2 = 300 000	Software limit.

Algorithm Constant	Description	Limit	Notes
		Common Server = 300 000	
IPL_{SL}[type_index]	Internet phone limit	5000	Software limit.
IPC_{HL}[type_index]	Internet phone calls per hour limit	CP PM = 40 000 CP DC = 80 000 COTS1 = 60 000 COTS2 = 80 000 Common Server = 80 000	Hardware limit.
PDmax[type_index]	Maximum number of PD users	CP PM = 15 000 CP DC = 25 000 COTS1 = 22 500 COTS2 = 40 000 Common Server = 40 000	Hardware limit.
$HVTC_{HL}$[type_index]	H.323 Gateway calls per hour limit	CP PM = 40 000 CP DC= 60 000 COTS1 = 60 000 COTS2 = 80 000 Common Server = 80 000	Hardware limit.
HVT_{SL}[type_index]	H.323 Gateway access ports for each Signaling Server	1200	CPU limit.
$SVTC_{HL}$[type_index]	SIP Gateway calls per hour limit	CP PM = 40 000 CP DC = 120 000 COTS1 = 60 000 COTS2 = 120 000 Common Server = 120 000	Hardware limit.
SVT_{SL}[type_index]	SIP Gateway access ports per Signaling Server	CP PM = 1800 CP DC = 3700 COTS1 = 1800 COTS2 = 3700 Common Server = 3700	CPU limit.
$TR87_{CL}$[type_index]	SIP CTI/TR87 clients	5000	CPU limit.
$SIPLC_{HL}$[type_index]	SIP Phone call per hour limit	CP PM = 15 000 CP DC =60 000 COTS1 = 25 000 COTS2 = 60 000 Common Server = 60 000	Hardware limit.

Algorithm Constant	Description	Limit	Notes
$SIPL_{SL}$[type_index]	SIP Phone limit	CP PM = 1800 CP DC = 3700 COTS1 = 1800 COTS2 = 3700 Common Server = 3700	Software limit, includes SIP DECT
$SIPL\_Vtrk_{SL}$[type_index]	Combined SIPL and Vtrk limit	CP PM = 1200 CP DC = 1600 COTS1 = 1600 COTS2 = 2000 Common Server = 2000	Software limit.
IPL_{DB}[type_index]	IP Phone limit with PD/CL/RL database and/or SIP Line	CP PM = 1500 CP DC = 2000 COTS1 = 2000 COTS2 = 3000 Common Server = 3000	Reduced due to PD/RL/CL database and other applications.
VT_{SL}[type_index]	Virtual Trunk share limit	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 1200 Common Server = 1200	VT share limit. H.323 + SIP.
$TR87_{SL}$[type_index]	SIP CTI/TR87 share limit	1000	Shared limit.
NRE_{SL}[type_index]	NRS endpoint share limit	100	Shared limit.
NRD_{SL}[type_index]	NRS routing entry share limit	1000	Shared limit.
$STBC_{HL}$[type_index]	SIP Trunk Bridge calls per hour	CP PM = 0 CP DC = 40 000 COTS1 = 0 COTS2 = 75 000 Common Server = 75 000	Hardware limit. CP PM, COTS1 not supported.
$STBC\_MA_{HL}$[type_index]	SIP Trunk Bridge calls per hour with Media Anchoring	CP PM = 0 CP DC = 8000 COTS1 = 0 COTS2 = 15 000 Common Server = 15 000	Hardware limit. CP PM, COTS1 not supported.

Algorithm Constant	Description	Limit	Notes
STB_{SL}[type_index]	SIP Trunk Bridge session limit	CP PM = 0 CP DC = 5000 COTS1 = 0 COTS2 = 5000 Common Server = 5000	Software limit. CP PM, COTS1 not supported.
STB_MA_{SL}[type_index]	SIP Trunk Bridge Media Anchoring session limit	CP PM = 0 CP DC = 1000 COTS1 = 0 COTS2 = 1000 Common Server = 1000	Software limit. CP PM, COTS1 not supported.
$MSCS_{SL}$[type_index]	MSC session limit	CP PM = 1800 CP DC = 4000 COTS1 = 1800 COTS2 = 4000 Common Server = 4000	Software limit. Sum of IPCONF + IPMUSIC + IPTONE + IPRAN + IPATTN dedicated platform
$MSCS_{SL}$IPCONF[type_index]	MSC IP Conf session limit	1920	Software limit dedicated platform
$MSCS_{SL}$IPRAN[type_index]	MSC IP Ran session limit	CP PM = 1000 CP DC = 1000 / 4000 COTS1 = 1000 COTS2 = 4000 Common Server = 4000	Software limit dedicated platform CP DC variance is 2GB / 4GB
$MSCS_{SL}$IPTONE[type_index]	MSC IP Tone session limit	CP PM = 1000 CP DC = 1000 / 4000 COTS1 = 1000 COTS2 = 4000 Common Server = 4000	Software limit dedicated platform CP DC variance is 2GB / 4GB
$MSCS_{SL}$IPMUSIC[type_index]	MSC IP Music session limit	CP PM = 1000 CP DC = 1000 / 4000 COTS1 = 1000 COTS2 = 4000 Common Server = 4000	Software limit dedicated platform CP DC variance is 2GB / 4GB

Algorithm Constant	Description	Limit	Notes
$MSCS_{SL}$IPATTN[type_index]	MSC IP Attn session limit	256	Software limit dedicated platform
$MSCC_{HL}$[type_index]	MSC session call rate limit	CP PM = 40 000 CP DC = 80 000 COTS1 = 60 000 COTS2 = 120 000 Common Server = 120 000	Hardware limit, calls per hour.
MSC_Vtrk_{SL}[type_index]	Shared MSC sessions and Vtrk limit	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 2000 Common Server = 2000	Software limit. Sum of IPCONF + IPMUSIC + IPTONE + IPRAN + IPATTN non-dedicated platforrm
MSC_Vtrk_{SL}IPCONF[type_index]	MSC IP Conf session limit	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 1000 Common Server = 1000	Software limit non-dedicated platform
MSC_Vtrk_{SL}IPRAN[type_index]	MSC IP Ran session limit	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 1000 Common Server = 1000	Software limit non-dedicated platform
MSC_Vtrk_{SL}IPTONE[type_index]	MSC IP Tone session limit	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 1000 Common Server = 1000	Software limit non-dedicated platform
MSC_Vtrk_{SL}IPMUSIC[type_index]	MSC IP Music session limt	CP PM = 800 CP DC = 1000 COTS1 = 1000 COTS2 = 1000 Common Server = 1000	Software limit non-dedicated platform
MSC_Vtrk_{SL}IPATTN[type_index]	MSC IP Attn session limit	256	Software limit non-dedicated platform
$MASS_{SL}$[type_index]	MAS session limit	CP DC = 240	Software limit

Algorithm Constant	Description	Limit	Notes
		IBM x3350 = 800 DELL R300 = 700 HP DL360-G7 = 800	CP PM and COTS1 not supported
$MASC_{SL}$[type_index]	MAS session call rate limit	CP DC = 6000 IBM x3350 = 18000 Dell R300 = 18000 HP DL360-G7 = 18000	Hardware limit for cph CP PM and COTS1 not-supported
Co-resCR[type_index]	Co-resident call rate limit	CP PM = 15 000 CP DC = 30 000 COTS1 = 30 000 COTS2 = 50 000 Common Server = 50 000	Call rate limit for Co-resident SS applications.
type_index = CP PM or CP DC or COTS1 (HP DL320-G4, IBM x306m), COTS2 (IBM x3350, Dell R300), or Common Server (HP DL360-G7) If a limit does not specify a type_index, then the limit applies to all the Signaling Server hardware platforms. CP MG is not supported as a stand-alone Signaling Server platform.			

Table 60: Signaling Server algorithm user inputs on page 277 describes the user inputs you use in the Signaling Server algorithm.

Table 60: Signaling Server algorithm user inputs

Algorithm user input	Description	Value	Notes
NRS[type_index]	Network Routing Service (NRS) required	enter	Yes or No
NRA[type_index]	Network Routing Service (NRS) alternate required	enter	Yes or No.
NRC[type_index]	NRS calls per hour	enter	Two components: $NRC = NRC_0 + NRC_{NET}$ one local, one network
NRD[type_index]	NRS CDP + UDP entries	enter	
NRE[type_index]	NRS endpoints	enter	(= 0 if NRS, which is a network-wide resource, is not

Algorithm user input	Description	Value	Notes
			provisioned in this node)
SSDB[type_index]	PD/RL/CL required	enter / derived	= a if not required = b if shared = c if standalone
IPL[type_index]	IP Phones	enter	
HVT[type_index]	Number of H.323 virtual trunks required	enter	
SVT[type_index]	Number of SIP virtual trunks required	enter	
TR87[type_index]	Aggregate number of SIP CTI/TR87 required based upon the MCS and OCS calculated	enter	
TR87A[type_index]	SIP CTI/TR87 redundancy required	enter	Yes or No
SIPL[type_index]	SIP Phones	enter	SIPN + SIP3
TPSA[type_index]	TPS N+1 redundancy required	enter	Yes or No
C_{UIP}	IP Phones calls per hour	enter / derived	Busy hour calls from all IP Phones
C_{UIP}	SIP Phones calls per hour	enter / derived	Busy hour calls from all SIP Phones
GWA[type_index]	H.323 Gateway alternate required	enter	Yes or No
GSA[type_index]	SIP Gateway alternate required	enter	Yes or No
SLGA[type_index]	SLG 1+1 redundancy required	enter	Yes or No
STB[type_index]	SIP Trunk Bridge sessions	enter	Requested SIP Trunk Bridge sessions
STB_MA[type_index]	SIP Trunk Bridge Media Anchoring required	enter	Yes or No

Algorithm user input	Description	Value	Notes
STBA[type_index]	STB 1+1 redundancy required	enter	Yes or No
MSCA[type_index]	MSC 1+1 redundancy required	enter	Yes or No
MASA[type_index]	MAS N+1 redundancy required	enter	Yes or No
XMSC_sessions	External MSC sessions from other systems	enter	0 if no external sessions expected
HS_NRSA[type_index]	HS NRS 1+1 redundancy required	enter	Yes or No
HS_ManA[type_index]	HS Manager 1+1 redundancy required	enter	Yes or No
HS_Primary	HS Primary location	enter	Yes or No, set for HS Primary system only

Table 61: Signaling Server algorithm variables on page 279 describes the variables you use in the Signaling Server algorithm.

Table 61: Signaling Server algorithm variables

Algorithm variable	Description	Value	Notes
NRP	NRS product of endpoing and CDP/UDP entries	-	Interim calculation.
C_{UIP}	UNIStim IP Phones calls per hour	calc	Busy hour calls from all UNIStim IP Phones.
HC_{VT}	H.323 call rate	calc	
SC_{VT}	SIP call rate	calc	
$ELCC_{VT}$	ELC call rate	calc	
C_{SIP}	SIP Phones calls per hour	calc	Busy hour calls from all SIP Phones.
SSNR	NRS Signaling Server calculation	calc	Real number requirement (example, 1.5) (= 0 if NRS is not provisioned in this node)

Algorithm variable	Description	Value	Notes
SSGW	NRS Signaling Server requirements	calc	Whole number requirement including alternate.
SSHR	H.323 Gateway Signaling Server calculation	calc	Real number requirement (example, 1.5).
SSHW	H.323 Gateway Signaling Server requirements	calc	Whole number requirement including alternate.
SSTR	TPS Signaling Server calculation	calc	Real number requirement (example,1.5).
SSTW	TPS Signaling Server requirements	calc	Whole number requirement including alternate.
SSTR87W	SIP CTI/TR87 Signaling Server requirements	calc	Whole number required including alternate.
SSTR87	SIP CTI/TR87 calculation	calc	Real number requirement.
SSSR	SIP Gateway Signaling Server calculation	calc	Real number requirement (example,1.5).
SSSW	SIP Gateway Signaling Server requirements	calc	Whole number required including alternate.
SSSLGR	SIP Line Gateway Signaling Server calculation	calc	Real number requirement (example,1.5).
SSSLGW	SIP Line Gateway Signaling Server requirements	calc	Whole number required including alternate.

Table 62: Constant and variable definitions for Co-resident Call Server and Signaling Server on page 281describes the constant and variable definitions for each Co-resident Call Server and Signaling Server hardware type.

Table 62: Constant and variable definitions for Co-resident Call Server and Signaling Server

Algorithm constant	Description	Value [type_index]	Comments
CS_SS_NRS_EP	Gateway endpoints on NRS	5	
CS_SS_NRS_RE	Routing entries on NRS	20	
CS_SS_CallRate	CS and NRS call rate limit	CP PM = 10 000 CP DC = 20 000 CP MG 128 = 8000 COTS2 = 30 000 Common Server = 30 000	
CS_SS_IP_limit	UNIStim Phone limit	CP PM = 1000 CP DC = 1000 CP MG 128 = 700 COTS2 = 1000 Common Server = 1000	
CS_SS_SIPL_limit	SIPL Phone limit	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 1000 Common Server = 1000	Includes SIP DECT
CS_SS_IP_SIPL_limit	UNIStim limit wien SIPL > 0	CP PM = 1000 CP DC = 1000 CP MG 128 = 700 COTS2 = 1000 Common Server = 1000	SIPL includes SIP DECT
CS_SS_Vtrk_limit	Vtrk limit	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	SIP Gateway + H323 Gateway
CS_SS_SIPL_Vtrk_limit	Combined limit of SIPL and Vtrk	CP PM = 800 CP DC = 800 CP MG 128 = 800 COTS2 = 1400 Common Server = 1400	sum of CS_SS_SIPL_limit + CS_SS_Vtrk_limit
CS_SS_UCM_Elements	UCM Element limit	CP PM = 10	

Algorithm constant	Description	Value [type_index]	Comments
		CP DC = 10 CP MG 128 = 10 COTS2 = 10 Common Server = 10	
CS_SS_MSC_Sessions	Maximum MSC sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	
CS_SS_MSC_Ses_Vtrk	Maximum combined Vtrk and MSC sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	MSC sessions + Vtrks <= this limit
CS_SS_MSC_IPCONF_Sessions	Maximum MSC IP Conf sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	
CS_SS_MSC_IPTONE_Sessions	Maximum MSC IP Tone sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	
CS_SS_MSC_IPRAN_Sessions	Maximum MSC IP Ran sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	
CS_SS_MSC_IPMUSIC_Sessions	Maximum MSC IP Music sessions	CP PM = 400 CP DC = 400 CP MG 128 = 400 COTS2 = 400 Common Server = 400	
CS_SS_MSC_IPATTN_Sessions	Maximum MSC IP Attn sessions	CP PM = 256 CP DC = 256 CP MG 128 = 256 COTS2 = 256	

Algorithm constant	Description	Value [type_index]	Comments
		Common Server = 256	
CS_SS_chasislimit	Chassis limit as a main	CP PM = 5 CP DC = 5 CP MG 128 = 5 COTS2 = 5 Common Server = 5	not used in CPU calculation
CS_SS_ACD_agents	Maximum ACD agents as a main	CP PM = 200 CP DC = 200 CP MG 128 = 200 COTS2 = 200 Common Server = 200	not used in CPU calculation
CS_SS_PRI	Maximum PRI spans	CP PM = 16 CP DC = 16 CP MG 128 = 16 COTS2 = 16 Common Server = 16	not used in CPU calculation
CS_SS_TDM	Maximum TDM Phones	CP PM = 720 CP DC = 720 CP MG 128 = 720 COTS2 = 720 Common Server = 720	not used in CPU calculation
CS_SS_branchs	Maximum Branch Offices	CP PM = 5 CP DC = 5 CP MG 128 = 5 COTS2 = 5 Common Server = 5	not used in CPU calculation
For Co-res CS and SS, type_index = CP PM, CP DC, CP MG 128, COTS2 and Common Server A type_index is required for each algorithm constant. If only one value is provided, it applies to all types COTS1 is not a supported hardware type for Co-res CS and SS. CP MG 32 is not supported as a stand-alone Co-res CS and SS. CP MG 32 is only supported as SIP Media Gateway.			

Signaling Server calculations

Signaling Server software components can run on shared or on standalone Signaling Servers. System traffic and user requirements (for alternate, redundant, or dedicated Signaling Servers) determine how many Signaling Servers are required.

SIP Dect telephones are provisioned on the SIP Line Gateway (SLG). The SIP Line count includes SIP Dect telephones (SipN + Sip3), where SipN includes both SIP Line and SIP Dect.

SIP Trunk Bridge requires a dedicated stand-alone multi-core Signaling Server (CP DC, COTS2, or Common Server),

SIP Vtrk use for the HMS400 application is not calculated here. The HMS400 server SIP Vtrk cannot be shared with a typical vtrk deployment on the NRS.

The Signaling Server algorithm takes account of all the requirements by performing the following calculations in sequence:

1. Avaya Aura Session Manager Servers on page 285
2. High Scalability system Signaling Servers on page 286
3. Signaling Server for Co-resident Call Server and Signaling Server on page 287
4. Signaling Server for Personal Directory/Callers List/Redial List database (SSDB) on page 288
5. Network Routing Service calculation (SSNR) on page 289
6. Terminal Proxy Server calculation (SSTR) on page 291
7. H.323 Gateway calculation (SSHR) on page 292
8. SIP Gateway calculation (SSSR) on page 293
9. SIP CTI/TR87 Calculation on page 293
10. SIP Line Gateway (SLG) calculation on page 294
11. Media Session Controller (MSC) calculation on page 295
12. Signaling Server Co-resident calculations on page 296
13. SIP DECT Server calculation on page 300
14. UCM calculation on page 300
15. SIP Trunk Bridge (STB) calculation on page 300
16. MAS calculation on page 301
17. Total Signaling Servers (SST) on page 304

Various hardware platforms are available for a Signaling Server. For the calculations, each variable is indexed by the Signaling Server type_index. The type_index = CP PM or CP DC or

COTS1 (HP DL320-G4, IBM x306m), COTS2 (IBM x3350, Dell R300), or Common Server (HP DL360-G7).

Note:

The Aura® Session Manager does not run on any of these severs and requires its own Aura® hardware.

1. **Avaya Aura® Session Manager Servers**

   A Session Manager (SM) can replace the NRS except for when IPv6, H.323 Gateway, or a High Scalability system is required.

   The Aura® Session Manager does not run on any of the CS 1000 supported signaling servers. It runs on its own Aura® supported platform. The calculation provided here is to help determine the number of Aura® Session Manager servers required. These are separate from the CS 1000 Signaling Server calculations and are not part of the total Signaling Server calculation.

   Once you determine if an Aura® Session Manager is required, the following table provides the information required for the Session Manager calculations.

   Table 63: Session Manager constants and variables

Algorithm constant	Description	Value	Notes
SME_EP_limit	SM maximum number of SIP endpoints	11 200	hardware limit for SM
SMC	SM calls per hours	enter	SMC =SMC_0 + SMC_{NET}, one local, one network equivalent to BHCC
SMC_{HL}	SM calls per hour	300 000	hardware limit for SM
SME	SM number of SIP endpoints	enter	Value from index 1h from user
SMNW	Number of SM	calc	

   SME is the sum of the value entered for index 1h and index 1d-1. The value entered for SME cannot exceed 100 000. The value calculated for SMC or BHCC cannot exceed 3 000 000, as the total number of SM is 10.

   SMC is the calculated value of the BHCC for the SM.

   $SMC_0 = VT_{SIP} \times CCS \times 100 \div WAHT$

   $SMC_{NET} = VTSM_{NET} \times CCS \times 100 \div WAHT \div 2$

Perform the following calculation to determine the number of Session Manager servers excluding N+1 redundancy or extra server requirements.

SMNR = larger of:

{

a) SME / SME_EP_limit — endpoints software limit

b) SMC(BHCC) / SMC_{HL} — calls per hour limit

c) 0 if SM not provisioned

}

If SMNR >= 1

Th
en {

SMNW = ROUNDUP (SSMR) if true; else 1

}

All Signaling Server calculations are required. Calculate the following algorithm in sequence

2. **High Scalability system Signaling Servers**

Determine if you require a High Scalability (HS) system. The HS system is a collection of High Availability (HA) Call Servers and Signaling Servers. Each HS system requires:

- a minimum of one dedicated COTS2 or Common Server NRS configured for SIP Proxy mode to handle PSTN and interoperability connections.
- a minimum of one dedicated COTS2 or Common Server for the High Scalability management server

Avaya recommends you deploy another inter-node NRS for management, and recommends you dedicate a UCM server at the primary location. Use the following calculation to determine the unique HS Signaling Servers required

If (HS_Primary = true)

Then {

HS_SS[type_index] = 2; (one NRS, one HS manager)

If (HS_NRSA = yes) then %alternate NRS required

HS_SS[type_index] = HS_SS[type_index] + 1

If (HS_ManA = yes) then %alternate HS manager required

HS_SS[type_index] = HS_SS[type_index] +1

```
        }
Else
        HS_SS[type_index] = 0;
```

All Signaling Server calculations are required. Calculate the following algorithm in sequence

3. **Signaling Server for Co-resident Call Server and Signaling Server**

Determine if you require a Co-resident Call Server and Signaling Server system. If yes, determine if the limits meet your Co-resident Call Server and Signaling Server limits. Provision a system only if the limit criteria is met.

```
If system_type = Co-res CS and SS % Various hardware platforms
Then
{
% Check for boundaries
Total_IP_Phones = UNIStim + SIP Line (including SIP Dect)
If (Total_IP_Phones > CS_SS_SIPL_limit[Co-res_CS_type])
        OR (SIPL_users > CS_SS_SIPL_limit[Co-res_CS_type])
        OR ((HVT + SVT + ELC_VT) > CS_SS_Vtrk_limit[Co-res_CS_type])
        OR ((HVT + SVT + MSC) > CS_SS_MSC_Ses_Vtrk[Co-res_CS_type])
        OR (MSC > CS_SS_MSC_sessions[Co-res_CS_type]) then
        {
        error (('CS-SS Co-res limit(s) exceeded.')
        Exit this section and select new system type;
        }
        Else % Set & Trunk limits OK, check NRS
If NRE > 0 and NRE <= CS_SSNRS_EP[Co-res_CS_type] then
If NRD <= CS_SS_RNS_RE[Co-res_CS_type] then
{
        calculate NRC as shown in NRS calculations;
        % Note that T_CALL is calculated in Call Server system calls section
        if T_CALL + NRC > CS_SS_CallRate[Co-res_CS_type] then
        {
        error ("The call rate calculated exceeds the limit this server can
        handle.");
```

Exit this section and select new system type;

}

Else

one Co-res_CS_type server provision is correct

}

Else

If (NRE > CS_SS_NRS_EP[Co-res_CS_type])

OR (NRD > CS_SS_NRS_RE[Co-res_CS_type]) then

{

If T_{CALL} > CS_SS_CallRate[Co-res_CS_type] then

{

error ("The call rate calculated exceeds the limit this server can handle.");

Exit this section and select a new system type;

}

Else % stand-alone NRS required, one of the Co-res NRS limits exceeded

SST[type_index] = SST[type_index] + NRA (=2 if true, else 1);

} % End of Co-res NRS bounds checking

% Provision additional SS for SIP Line if required

If SIP Line users > 0 and SIP Line dedicated= YES then

if SIPLineRedundant = true then

SST[type_index] = SST[type_index] + 2

Else

SST[type_index] = SST[type_index] + 1

} % End of Co-res CS + SS checks

Else

All Signaling Server calculations are required. Calculate the following algorithm in sequence

4. **Signaling Server for Personal Directory/Callers List/Redial List database (SSDB)**

SSDB[type_index]	=	a	if no PD/CL/RL feature

= b if yes on feature, and sharing functions on SS (UNIStim Phones <= IPL_{DB}[type_index] and (HVT+SVT) <= VT_{SL}[type_index]); PD_Co-res = True; NumOfCo-res = NumOfCo-res + 1;

= c if yes on feature, and dedicated PD/CL/RL or (UNIStim Phones > IPL_{DB}[type_index] and (HVT+SVT) > VT_{SL}[type_index])
if SSDB = c PD on dedicated Signaling Server
SST[type_index] = SST[type_index] +1

5. **Network Routing Service calculation (SSNR)**

Choose NRS hardware type if dedicated NRS, otherwise nrs_type_index = type_index

If dedicated NRS required, then

{ obtain the NRS platform type: nrs_type_index

where nrs_type_index = CP PM or CP DC or COTS1 (IBM x306m, HP DL320-G4), COTS2 (IBM x3350, Dell R300), or Common Server (HP DL360-G7)

initialize SST[nrs_index_type] = 0

}

If there are SIP endpoints that require NRS, query and obtain the SIP mode,

If SIP_mode = none, Proxy or Redirect.

If SIP_mode = Proxy, then the maximum number of GW endpoints = 1000 (SIP_Proxy_limit).

Query and obtain SIP transport required: SIP_transport = UDP or TCP/TLS

If SIP_mode = Proxy and SIP_transport = TCP/TLS, then

NRS_EP_limit = SIP_Proxy_limit[nrs_type_index]

Else

NRS_EP_limit = NRE[nrs_type_index]

Since the capacity for handling H323 calls is different than SIP calls, you must determine the SIP call loading factor on NRS. There are two SIP modes, SIP_Proxy and SIP_Redirect. To calculate the SIP loading factor, see Table 64: SIP mode factors on page 289.

Table 64: SIP mode factors

type_index	SIP_Redirect_Factor	SIP_Proxy_Factor
CP PM	2	4

type_index	SIP_Redirect_Factor	SIP_Proxy_Factor
CP DC	1.5	3
COTS1	1.5	3
COTS2	1.67	2.5
Common Server	1.67	2.5

SSNR[nrs_type_index] = larger of:

{

a NRE[nrs_type_index] ÷ NRS_EP_limit (endpoints software limit)

b NRD[nrs_type_index] ÷ NRD_I[nrs_type_index] (dial plan entries software limit)

c NRC[nrs_type_index] ÷ 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_type_index] or NRD > NRD_{SL}[nrs_type_index] or NRC > NRC_{SL}[nrs_type_index]), then 1, else 0 (Co-res limit)

}

If you require a dedicated NRS Signaling Server, round up SSNR for the following calculations.

```
If SSNR[nrs_type_index] >= 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[nrs_type_index];
If SSNR[type_index] > 0
Then {  NRS_Co-res = True; NumOfCo-res = NumOfCo-res + 1
        }
        }
```

NRC could be a hardware or CPU or memory limit; it includes NRC_0 (calls result from main switch calculation) and network VT_{NET} for the Network Routing Service: $NRC = NRC_0 + NRC_{NET}$

Both VT_{323} and VT_{SIP} must convert to H.323 and SIP calls from your input:

H323 calls = VT_{323} × CCS per VT × 100 ÷ WAHT

SIP calls = VT_{SIP} × CCS per VT × 100 ÷ WAHT + (ELC_{VT} × CSS per telephone × 100 ÷ WAHT)

Determine the SIP loading factor on the NRS:

Factor = If SIP mode = Proxy,

Then SIP_Proxy_Factor[nrs_type_index]

Else If SIP mode = Redirect,

Then SIP_Redirect_Factor[nrs_type_index]

Else 0

NRC_0 = (H323 calls + Factor × SIP calls) NRC_{NET} = VT_{NET} × CSS for each VT × 100 ÷ WAHT ÷ 2

NRC = NRC_0 + NRC_{NET}

Formula (c) in SSNR equation = NRC ÷ NRC_{HL}[nrs_type_index]

The previous equation represents the load on the Signaling Server to handle NRS calls. Compare it with (a) and (b) to determine the highest of all potential uses.

6. **Terminal Proxy Server calculation (SSTR)**

Calculate the TPS call rate: C_{UIP}= $C2_{IP}$× 2 + $C1_{IP}$ + $C2_{SIPUIP}$ + C_{STIV} + C_{STID} + C_{STVI}

The Call Server CPU calculations define the variables.

SSTR[type_index] = larger of:

{

a	IPL ÷ IPL_{SL}[type_index]	IP Phones software limit
b	C_{UIP}[type_index] ÷ IPC_{HL}[type_index]	call limit - calls per hour limit
c	If IPL > IPL_{DB}[type_index] then 1 else 0	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[type_index] >= 1 or dedicated TPS required

Then {

SSTW[type_index] = ROUNDUP(SSTR[type_index]) + TPSA[type_index] (=1, if true; else 0)

TPSA = if N+1 redundant TPS needed

SST[type_index] = SST[type_index] + SSTW;

}

Else { Co-res

If SSTR[type_index] = 0

Then { TPS_Co-res = true; NumOfCo-res = NumOfCo-res + 1

}

}

7. **H.323 Gateway calculation (SSHR)**

$HC_{VT} = (HVT_{CCS} \times 100) \div WAHT$

SSHR[type_index] = larger of:

{

a	HVT[type_index] ÷ HVT_{SL}[type_index]	number of trunks (software limit)
b	HC_{VT}[type_index] ÷ $HVTC_{HL}$[type_index]	calls per hour (hardware limit)
c	If HVT[type_index] > VT_{SL}[type_index] then 1 else 0	non-dedicated limit

}

If the user wants H.323 Gateway(s) in a dedicated Signaling Server, round up SSHR[type_index] before proceeding with further calculations:

if SSHR[type_index] >= 1 or dedicated H323 Gateway is required

Then {

SSHW[type_index] = ROUNDUP(SSHR[type_index]) × GWA[type_index] (true = 2; else=1)

GWA = If Alternate H323 Gateway needed

SST[type_index] = SST[type_index] + SSSW

}

Else { If SSHR[type_index] > 0

Then { H323_Co-res = True; NumOfCo-res = NumOfCo-res + 1

}

}

8. **SIP Gateway calculation (SSSR)**

   The number of SIP virtual trunk calls is calculated from the SVT_{CSS}:

   $SC_{VT} = (SVT_{CCS} \times 100) \div WAHT$

   The number of virtual trunks required for the Extend Local Calls (ELC) feature:

   ELC_{VT} = ELC ISM value (the requested number of ELC users)

   The number of ELC calls that impact SIP virtual trunks:

   $ELCC_{VT} = C_{SS} \times ELC\_P$

   SSSR[type_index] = larger of:

   {

a	(SVT[type_index] + ELC_{VT}) ÷ SVT_{SL}[type_index]	number of trunks (software limit)
b	(SC_{VT}[type_index] + $ELCC_{VT}$	calls per hour (hardware limit)
c	(SVT[type_index]+ ELC_{VT}) > VT_{SL}[type_index] then 1 else 0	non-dedicated 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[type_index] >= 1 or dedicated SIP Gateway is required

   Then {

   SSSW[type_index] = ROUNDUP(SSSR[type_index]) + GSA[type_index] (= 2 if true, else 1)

   GSA = If Alternate SIP Gateway needed

   SST[type_index] = SST[type_index] + SSSW

   }

   Else { If SSSR[type_index] > 0

   Then { SIP_Co-res = true; NumOfCo-res = NumOfCo-res + 1

   }

   }

9. **SIP CTI/TR87 Calculation**

   If SIP CTI TR87 feature is present, Total SIP CTI TR87 is > 0

   SSTR87[type_index] = larger of:

{

a	TR87[type_index] ÷ $TR87_{CL}$[type_index]	number of clients (software limit)
b	If TR87 > $TR87_{SL}$[type_index] then 1 else 0	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[type_index] >= 1 or dedicated SIP CTI is required

Then {

SSTR87W[type_index] = ROUNDUP(SSTR87[type_index]) × TR87A[type_index] (= 2 if true, else 1)

TR87A = if Alternate SIP CTI/TR87 needed

SST[type_index] = SST[type_index] + SSTR87W

}

Else { If SSTR87[type_index] > 0

Then { TR87_Co-res = true; NumOfCo-res = NumOfCo-res + 1

}

}

10. **SIP Line Gateway (SLG) calculation**

SLG supports SIP DECT as SIP Line users.

The number of SIP Line users on a Call Server is defined as: SIPL = SIPN + SIP3 Where SIPN = ISM value including SIP Line and SIP DECT telephones

If SIPL = 0 do not provision hardware for SIPL.

Calculate the total number of SIP Line calls

$C_{SIP} = (2 \times C_{2SIP}) + C_{1SIP} + C_{2SIPUIP} + C_{STSV} + C_{STSD} + C_{TSVS} + C_{TSDS}$

The Call Server CPU calculations define the variables

SSSLGR7[type_index] = larger of:

{

a	SIPL ÷ $SIPL_{SL}$[type_index]	number of phones (software limit)
b	C_{SIP} ÷ $SIPLC_{HL}$[type_index]	calls per hour (software limit)

c	SIPL + IPL > IPL_db or SIPL + SVT + HVT > $SIPL\_Vtrk_{SL}$[type_index] then 1 else 0	non-dedicated limit

}

Round up SSSLGR before performing further calculations

If (SSLGR[type_index] >= 1) or dedicated SIP Line Gateway then

{

SSSLGW[type_index] = ROUNDUP (SSSLG[type_index] × SIPLA[type_index]

(=2, if true; else 1) SIPLA = if 1 + 1 redundant SIPL is required

SST[type_index] = SST[type_index] + SSSLGW;

}

Else — Co-res

If SSSLGR[type_index] > 0 then

{

SLG_Co-res = true;

NumOfCo-res = NumOfCo-res + 1;

}

11. **Media Session Controller (MSC) calculation**

The MSC application is required if you selected IP Media Services and calculated the required media sessions. To calculate MSC_sessions, see IP Media Services on page 222.

MSCC = (MSC_sessions × TRK_{CCS} × 100) ÷ WAHT

Media anchoring lowers the supported number of sessions and the call rate of SIP Trunk Bridge.

SSMSCR[type_index] = larger of:

{

a	MSC_sessions ÷ $MSCS_{SL}$[type_index]	number of sessions (software limit)
b	MSCC ÷ $MSCC_{SL}$[type_index]	calls per hour (hardware limit)
c	MSC_sessions + SVT + HVT > MSC_Vtrk_{SL}[type_index] then 1, else 0	1 = dedicated MSC required 0 = non-dedicated limit OK

```
}
SSMSCR[type_index] = ROUNDUP (SSMSCR[type_index]);
If (SSMSCR[type_index] >= 1) or dedicated MSC then
    {
    If MSCA[type_index] = true then              redundant MSC
                                                 requested
    SSMSCW[type_index] =
    SSMSCR[type_index] + 1
    SST[type_index] = SST[type_index] +
    SSMSCW[type_index];
    }
    Else                                         Co-res
    If SSMSCR[type_index] > 0 then
    {
    MSC_Co-res = true;
    NumOfCo-res = NumOfCo-res + 1;
    }
```

12. **Signaling Server Co-resident calculations**

    Determine if any Signaling Server applications can co-reside on one Signaling Server.

```
Case of NumOfCo-res;
Null;                                        No SS applications Co-
                                             res
{
SST[type_index] = SST[type_index] + 1;       One SS application -
                                             assign one SS
      If redundant needed, add one SS and reset Co-res flag
If (NRS_Co-res = true and NRA = true) or (TPS_Co-res = true and TPSA = true)
or (H323_Co-res = true and GWA = true) or (SIP_Co-res = true and GSA = true)
or (TR87_Co-res = true and TR87A = true) or MSC_Co-res = true and MSCA =
true)

Then {  SST[type_index] = SST[type_index] + 1;
        If NRS_Co-res = true then NRS_Co-res = false;
        Else
```

If TPS_Co-res = true then TPS_Co-res = false;

Else

If H323_Co-res = true then H323_Co-res = false;

Else

If SIP_Co-res = true then SIP_Co-res = false;

Else

If TR87_Co-res = true then TR87_Co-res = false;

Else

If SIPL_Co-res = true then SIPL_Co-res = false;

Else

If MSC_Co-res = true then MSC_Co-res = false;

NumOfCo-res = 0;

	}	End case of one SS application Co-res
}		
	Determine if the Co-res VTRK limit is exceeded	More than one SS application Co-res
{		

If H323_Co-res = true and SIP_Co-res = true and (HVT + SVT + ELC_{VT} > VT_{SL}[type_index])

Then { SST[type_index] = SST[type_index] + 1;

H323_Co-res = false;

NumOfCo-res = NumOfCo-res + 1;

If GWA = true then SST[type_index] = SST[type_index] + 1;

}

If NumOfCo-res = 0, then exit

Else If NumOfCo-res = 1, then DO One SS application - assign one SS

Else

{ CallRate = 0;

If NRS_Co-res = true then CallRate = CallRate + NRC;

If TPS_Co-res = true then CallRate - CallRate + C_{UIP};

If H323_Co-res = true then CallRate = CallRate + HC_{VT};

If SIP_Co-res = true then CallRate = CallRate + SC_{VT}+ $ELCC_{VT}$;

If SIPL_Co-res = true then CallRate = CallRate + C_{SIP};

If MSC_Co-res = true then CallRate = CallRate + MSCC;

If CallRate <= Co-resCR[type_index]

Then {

SST[type_index] = SST[type_index] + 1;

If (NRS_Co-res = true and NRA = true) or (TPS_Co-res = true and TPSA = true) or (H323_Co-res = true and GWA = true) or (TR87_Co-res = true and TR87A = true) or (SIPL_Co-res = true and SIPLA = true) or (MSC_Co-res = true and MSCA = true)

Then

SST[type_index] = SST[type_index] + 1;

}

Else {	Call rate exceeds the Co-res limit

TNRC = NRC;

TC_{UIP} = C_{UIP};

THC_{VT} = HC_{VT};

TSC_{VT} = SC_{VT} + $ELCC_{VT}$;

TC_{SIP} = C_{SIP};

TMSCC = MSCC

DO

{

large = largest of (TNRC, TC_{UIP}, THC_{VT}, TSC_{VT}, T_{SIP}, TMSCC);	Dedicate a SS to the largest contributor

If large = TNRC

Then {

NRS_Co-res = false;

CallRate = CallRate - NRC;

NumOfCo-res = NumOfCo-res - 1;

TNRS = 0;

SST[type_index] = SST[type_index] + NRA (= 2 if true, else 1);

}

```
Else      If large = TC_UIP
Then {    TPS_Co-res = false;
          CallRate = CallRate = C_UIP;
          NumOfCo-res = NumOfCo-res - 1;
          TC_UIP = 0;
          SST[type_index] = SST[type_index] + TPSA (= 2 if true, else 1);
}
Else      If large = THC_VT
Then {    H323_Co-res = false;
          CallRate = CallRate - HC_VT;
          NumOfCo-res = NumOfCo-res - 1;
          THC_VT = 0;
          SST[type_index] = SST[type_index] + GWA (= 2 if true, else 1);
}
Else      If large = TSC_VT
Then {    SIP_Co-res = false;
          CallRate = CallRate - SC_VT - ELCC_VT;
          NumOfCo-res = NumOfCo-res - 1;
          TSC_VT = 0;
          SST[type_index] = SST[type_index] + GSA (= 2 if true, else 1);
}
Else      If large = TC_SIP
Then {    SIPL_Co-res = false;
          CallRate = CallRate - SC_SIP;
          NumOfCo-res = NumOfCo-res - 1;
          TSC_SIP = 0
          SST[type_index] = SST[type_index] +
          SIPLA (=2 if true, else 1);
}
Else      If large = TMSCC
Then {    MSC_Co-res = false;
          CallRate = CallRate - MSCC;
          NumOfCo-res = NumOfCo-res - 1;
```

TMSCC = 0

SST[type_index] = SST[type_index] + MSCA (=2 if true, else 1);

}

Repeat DO until CallRate <= Co-resCR[type_index]

If NumOfCo-res = 0, then exit

Else If NumOfCo-res = 1, then do — One SS application - Assign one SS

Else { SST[type_index] = SST[type_index] + 1;

If (NRS_Co-res = true and NRA = true) or (TPS_Co-res = true and TPSA = true) or (H323_Co-res = true and GWA = true) or (SIP_Co-res = true and GSA = true) or (TR87_Co-res = true and TR87A = true) or (SIPL_Co-res = true and SIPLA = true) or MSC_Co-res = true and MSCA = true)

Then SST[type_index] = SST[type_index] + 1;

}

}

}

}

13. **SIP DECT Server calculation**

    The SIP Line Gateway (SLG) calculation includes SIP DECT. No independent SIP DECT servers required.

    SS_DECT = 0

14. **UCM calculation**

    If ucm_pss_required = true

    Then {

    SST[ucm_pss_type] = SST[ucm_pss_type] + 1;

    If ucm_backup_required = true then

    SST[ucm_pss_type] = SST[ucm}pss_type] + 1;

    }

15. **SIP Trunk Bridge (STB) calculation**

    The SIP Trunk Bridge application must be deployed as a stand-alone dedicated server.

Determine the number of calls per hour for SIP Trunk Bridge. STBC[type_index] = (STB[type_index] × TRK_{CCS} × 100) ÷ WAHT The number of servers required increases with media anchoring usage. Media anchoring lowers the supported call rate and number of sessions for SIP Trunk Bridge.

If STB_MA[type_index] = yes then		
{		Media anchoring required
SSSBTR[type_index] = larger of:		
{		
a	STB[type_index] ÷ STB_MA_{SL}[type_index]	number of trunks (software limit)
b	STBC[type_index] ÷ $STBC\_MA_{HL}$[type_index]	calls per hour (hardware limit)
}		
Else		No media anchoring
{		
SSSBTR[type_index] = larger of:		
{		
a	STB[type_index] ÷ STB_{SL}[type_index]	number of trunks (software limit)
b	STBC[type_index] ÷ $STBC_{HL}$[type_index]	calls per hour (hardware limit)
}		
SIP Trunk Bridge requires a dedicated Signaling Server		
If STBA[type_index] = no then		No redundancy required
	{	
	SSSTBW[type_index] = ROUNDUP (SSSR[type_index])	
	Else	
	SSSTBW[type_index] = ROUNDUP (SSSR[type_index]) + 1	
	}	
SST[type_index] = SST[type_index] + SSSTBW[type_index];		

16. **MAS calculation**

The MAS application requires additional dedicated Signaling Servers. The MAS application is supported on CPDC and COTS2 Server Signaling Servers only.

MAS servers are deployed in clusters, with a maximum cluster size of 7 servers.

The MAS license server does not know the capacity limitations of the MAS servers in the cluster. Therefore, all MAS servers within the cluster **MUST** be of the same type, or at a minimum, have the same capacity rating.

To ensure the MAS servers can handle the load of a cluster when one of the servers fails, a redundant server may be selected. Selecting redundancy for MAS puts an additional server in **each** cluster (N+1 servers per cluster).

Since the cluster has a limit of 7 servers, adding a redundant server lowers the cluster size by 1 server, which could potentially increase the number of MAS clusters required. An example is shown in the following table:

No Redundancy		**Redundancy**		
Servers in Cluster		Servers in Cluster	Redundant Server	Total Server is Cluster
1–7		1–6	1	2–7

Additional MAS servers requested by the customer must also be considered in the cluster calculations.

The **Systems Options** page contains an input field for Additional MAS Servers, referred to as **MAS_Additional**.

The codec used on the MAS server has a large impact on the number of conference sessions that can be supported. Therefore, the MAS codec selection must be taken into consideration when determining the number of sessions that can be supported on a MAS server.

The following table shows the MAS Codec Ratios (MASCRatio) used for each MAS server type:

Type index	G711	G722	G729
CPDC	1	1.8	1.944
Dell R300	1	1.8	1.944
IBM x3350	1	1.8	1.944
HP GL360 G7	1	1.8	1.944

The following table shows the MAS SRTP impact (MASSRTP):

Type index	G711	G722	G729
CPDC	0.860	0.922	0.928
Dell R300	0.860	0.922	0.928
IBM x3350	0.860	0.922	0.928
HP GL360 G7	0.860	0.922	0.928

The calculations used to determine the number of Signaling Servers required for MAS depends on the MAS codec selection, the use of Media Security on the MAS server and two MSC variables; MSC Sessions (**MSC_sessions**) and MSC call rate (**MSCC**).

If (MAS_MSEC=off) then		
	MASSRTP_factor = 1;	
Else		
	MASSRTP_factor = MASSRTP[codec];	
SSMASR[type_index] = larger of:		
{		
a	(MSC_sessions + XMSC_sessions) ÷ ((MASSSL[type_index] * MASSRTP_factor)/ MASCRatio[MAScodec])	number of sessions (software limit)
b	MSCC ÷ $MASC_{HL}$[type_index]	calls per hour (hardware limit)
}		
SSMASR[type_index] = ROUNDUP(SSMASR[type_index]) + MAS_Additional		
If (SSMASR [MAS_type_index] >= 1)		
Then		
{		
	If MASA[MASA_type_index] = true then	Redundant MAS Server(s) required
	{	
	SSMASRed[MAS_type_index] = SSMASR[MAS_type_index] / (MAS_Cluster_size -1)	Calculate number of redundant servers
	SSMASRed[MAS_type_index] = ROUNDUP (SSMASR[MAS_type_index])	Roundup to whole number

	SSMASW[MAS_type_index] = SSMASR[MAS_type_index] + SSMASRed[MAS_type_index]	Total MAS servers = N server + redundant server
	}	
Else	SSMASW[MAS_type_index] = SSMASR[MAS_type_index]	No redundancy. N is total MAS Servers required
}		
SST[MAS_type_index] = SST[MAS_type_index] + SSMASW[MAS_type_index]		Add total MAS servers to current SS total

17. **Total Signaling Servers (SST)**

    The total number of Signaling Servers provisioned:

    SST[type_index] = SST[type_index] + SST[nrs_type_index] + SST[ucm_pss_type] + SST[sipl_type_index]:

    If you are calculating for the primary location of a High Scalability (HS) deployment, then deploy the HS servers at this location:

    SST[type_index] = SST[type_index] + HS_SS[type_index]

See Signaling Server calculation on page 316 for a 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 ÷ RSPE) × [FR – (RFR_S or RFR_C)] × (DDR ÷ 24) × (RSP_C)

Simplified formulas:

RSF = (16 000 ÷ RSPE) for stand-alone Network Routing Service RSF = (10 000 ÷ RSPE) for collocated Network Routing Service

Table 65: RSF algorithm constant and variable definitions on page 305 defines the terms used in the calculations.

Table 65: 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)	
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 Enterprise Configurator 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 on page 306
- Line and trunk traffic on page 306

Virtual Trunks

When the VT number input by the user differs significantly from the calculated VT number (more than 20% difference), the Enterprise Configurator 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 Enterprise Configurator 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 Enterprise Configurator 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 on page 306). 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 Enterprise Configurator 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}) × WAHT ÷ 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}) × WAHT ÷ 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 66: Tips to reduce traffic imbalances on page 307 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 66: 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

When this happens...	Try this...
	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): 0.25
- Tandem ratio (R_T): 0.03
- Incoming ratio (I): 0.60
- Outgoing ratio (O): 0.12

  In fraction of calls, the above ratios add up to 1.
- AHT_{SS} = 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 Communication Server 1000E CP PIV system with the following configuration data:

- 1200 digital and analog telephones at 5 CCS/telephone
  - including 170 ACD agents with digital telephones at 33 CCS/agent
- 1600 IP telephones at 5 CCS/IP telephone
  - including 50 IP ACD agent telephones at 33 CCS/IP agent telephone
- 200 MDECT mobile phones at 5 CCS/telephone
- 1200 SIP Line telephones

- 820 trunks
    - 450 Virtual Trunks (300 H.323 and 150 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.)
    - 370 TDM (PRI) trunks at 28 CCS/trunk
- 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, preselected application ports that require DSP channels and real time feature overhead. The DSP required for IP Phones to access these special applications is proportional to the percentage of IP calls in the system.
    - 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)
    - BRI users: 8 (HT = 180)
    - MDECT mobile telephones: 200 (HT = WAHT)
- Features with processing overhead but no hardware ports:
    - CPND percentage: CPND calculation assumes all calls involving a telephone use CPND
    - Converged Desktop percentage: 5% of the following calls: (intraoffice calls × 0.1) + incoming calls + outgoing calls + tandem calls
    - Intraoffice CDR: No (could 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 53: Real time factors on page 258.

Calculations

The calculations in this example were performed by spreadsheet. Some rounding off may have occurred.

- The percentage of ACD agent to total telephones = (50 + 170) ÷ (1200 + 1600 + 1200 + 200) × 100 = 5.238 % This ratio is less than the 15% threshold, so the site is not considered a call center. All ACD traffic will be included in call distribution calculations. For more information, see DSP ports for general traffic on page 266. The following calculations use the default nonblocking telephone CCS rate of 18 CCS.
- L_{TDM} TDM telephones CCS = [(1200 – 170) × 5] + (170 × 18) = 8210 CCS
- L_{IP} IP telephones CCS = (1600 – 50) × 5 = 7750 CCS
  - L_{ACD} TDM ACD agent CCS = 170 × 33 = 5610 CCS
  - L_{ACDIP} IP ACD agent CCS = 50 × 33 = 1650 CCS
  - L_{DECT} DECT telephones CCS = 200 × 5 = 1000 CCS
  - L_{SIPL} SIP Line telephones CCS = 1200 × 5 = 6000 CCS
  - ACD_{adj} ACD CCS adjustment for TDM agents = 170 × 18 = 3060 CCS
  - L_{CCS} Total line CCS = 8210 + 7750 + 5610 + 1650 + 1000 + 6000 + 3060 = 27160 CCS
- T_{TDM} TDM trunk CCS = 370 × 28 = 10360 CCS
  - HVT_{CCS} H.323 trunk CCS = 300 × 28 = 8400 CCS
  - SVT_{CCS} SIP trunk CCS = 150 × 28 = 4200 CCS
  - VT_{CCS} Total Virtual Trunk CCS = 8400 + 4200 = 12600 CCS
  - T_{TCCS} Total Trunk CCS = 12600 + 10360 = 22960 CCS
- Fraction of H.323 CCS of total Virtual Trunk CCS (V_H) = 8400 ÷ 12600 = 0.67
- Fraction of SIP CCS of total Virtual Trunk CCS (V_S) = 4200 ÷ 12600 = 0.33
- Fraction of Virtual Trunk CCS of total trunk CCS (V) = 12600 ÷ 22960 = 0.549
- Fraction of UNIStim IP CCS (P_U) = (7750 + 1650) ÷ 27160 = 0.346
- Fraction of SIP CCS (P_S) = 6000 ÷ 27160 = 0.221
- Fraction of IP CCS (P_{IP}) = 0.346 + 0.221 = 0.561
- Weighted average holding time (WAHT) = (60 × 0.25) + (150 × 0.60) + (150 × 0.12) + (150 × 0.12) + (180 × 0.03) = 128 seconds
- CP1 local CallPilot CCS = 36 × 36 = 936
- CP2 network CallPilot CCS = 24 × 26 = 624
- Total CCS (T_{CCS}) = L_{CCS} + T_{TCCS} = 27160 + 22960 = 50120 CCS
- Total calls (T_{CALL}) = 0.5 × T_{CCS} × 100 ÷ WAHT = 0.5 × 50120 × 100 ÷ 128 = 19578

- 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}).

  a. Intraoffice calls (C_{SS}) = $T_{CALL} \times R_I$ = 19578 × 0.25 = 4895 calls

     i. Intraoffice UNIStim IP to UNIStim IP calls (C_{2IP}) = $C_{SS} \times P_U \times P_U$ = 4895 × 0.346 × 0.346 = 586 (require no DSP, no VT) P_UIPtoUIP = 586 ÷ 19578 = 0.03

     ii. Intraoffice UNIStim IP to TDM calls (C_{1IP}) = $C_{SS} \times 2 \times P_U \times (1 – P_U)$ = 4895 × 2 × 0.346 × (1 – 0.346) = 1467 (require DSP) P_UIPtoL = 1467 ÷ 19578 = 0.07

     iii. Intraoffice TDM to TDM calls (C_{NoIP}) = $C_{SS} \times (1 – P_{IP})^2$ = 4895 × (1 – 0.567) × (1 – 0.567) = 918 (require no DSP, no VT) P_LtoL = 918 ÷ 19578 = 0.05

     iv. Intraoffice SIP Line to SIP Line calls (C_{2sip}) = $C_{SS} \times P_S^2$ = 4895 × 0.221 × 0.221 = 239 (require no DSP, no VT) P_SIPtoSIP = 239 ÷ 19578 = 0.01

     v. Intraoffice SIP Line to UNIStim IP calls ($C_{2sipuip}$) = $C_{SS} \times P_S \times P_U$ = 4895 × 0.221 × 0.346 = 748 (require no DSP, no VT) P_SIPtoUIP = 748 ÷ 19578 = 0.04

     vi. Intraoffice SIP Line to TDM calls (C_{1sip}) = $C_{SS} \times 2 \times P_S \times (1 - P_{IP})$ = 4895 × 2 × 0.221 × (1 - 0.567) = 936 (require DSP, no VT) P_SIPtoL = 918 ÷ 19578 = 0.05

  b. Tandem calls (C_{TT}) = $T_{CALL} \times R_T$ = 19578 × 0.03 = 587 calls

     i. Tandem VT to TDM calls (C_{T1VT}) = 2 × C_{TT} × V × (1 – V) = 2 × 587 × 0.549 × (1 – 0.549) = 291 (require DSP and VT) P_VTtoTr = 291 ÷ 19578 = 0.0015

     ii. Tandem TDM to TDM calls (C_{T2NoVT}) = C_{TT} × (1 – V) × (1 – V) = 587 × (1 – 0.549) × (1 – 0.549) = 120 (require no DSP, no VT) P_TrtoTr = 120 ÷ 19578 = 0.006

     iii. Tandem VT (H.323) to VT (SIP) calls (C_{T2HS}) = $C_{TT} \times V^2 \times V_H \times V_S$ × 2 × 2 = 587 × 0.549 × 0.549 × 0.67 × 0.33 × 4 = 157 (require no DSP, VT) P_VhtoVs = 157 ÷ 19578 = 0.008

  c. Originating/outgoing calls (C_{ST}) = T_{CALL} × O = 19578 × 0.12 = 2349 calls

     i. UNIStim IP to VT calls (C_{STIV}) = $C_{ST} \times P_U$ × V = 2349 × 0.346 × 0.549 = 446 (require VT) P_UIPtoVT = 446 ÷ 19578 = 0.02

     ii. UNIStim IP to TDM trunk calls (C_{STID}) = $C_{ST} \times P_U$ × (1 – V) = 2349 × 0.346 × (1 – 0.549) = 367 (require DSP) P_UIPtoTr = 367 ÷ 19578 = 0.02

     iii. TDM telephone to VT calls (C_{STDV}) = $C_{ST} \times (1 – P_{IP})$ × (V) = 2349 × (1 – 0.567) × 0.549 = 558 (require DSP, VT) P_LtoVT = 558 ÷ 19578 = 0.03

iv. TDM to TDM calls (C_{STDD}) = C_{ST} × (1 – P_{IP}) × (1 – V) = 2349 × (1 – 0.567) × (1 – 0.549) = 459 (require no DSP, no VT) P_LtoTr = 459 ÷ 19578 = 0.02

v. SIP Line to VT calls (C_{STSV}) = C_{ST} × P_S) × V = 2349 × 0.221 × 0.549 = 285 (require no DSP, VT) P_SIPtoVT = 285 ÷ 19578 = 0.01

vi. SIP Line to TDM trunk calls (C_{STSD}) = C_{ST} × P_S) × (1 - V) = 2349 × 0.221 × (1 - 0.549) = 234 (require DSP, no VT) P_SIPtoTr = 234 ÷ 19578 = 0.01

d. Terminating/incoming calls (C_{TS}) = T_{CALL} × I = 19578 × 0.60 = 11747 calls

i. VT to TDM telephone calls (C_{TSVD}) = C_{TS} × V × (1 – P_{IP}) = 11747 × 0.549 × (1 – 0.567) = 2791 (require DSP, VT) P_VTtoL = 2791 ÷ 11747 = 0.14

ii. VT to UNIStim IP calls (C_{TSVI}) = C_{TS} × V × P_U = 11747 × 0.549 × 0.346 = 2231 (require VT) P_VTtoUIP = 2231 ÷ 11747 = 0.11

iii. TDM to UNIStim IP calls (C_{TSDI}) = C_{TS} × (1 – V) × P_U = 11747 × (1 – 0.549) × 0.346 = 1834 (require DSP) P_TrtoUIP= 1834 ÷ 11747 = 0.09

iv. TDM to TDM telephone calls (C_{TSDD}) = C_{TS} × (1 – V) × (1 – P_{IP}) = 11747 × (1 – 0.549) × (1 – 0.567) = 2295 (require no DSP, no VT) P_LtoL = 2295 ÷ 11747 = 0.12

v. VT to SIP Line calls (C_{TSVS}) = C_{TS} × V × P_S = 11747 × 0.549 × 0.221 = 1424 (require no DSP, VT) P_VTtoSIP = 2424 ÷ 11747 = 0.07

vi. TDM trunk to SIP Line calls (C_{TSDS}) = C_{TS} × (1 - V) × P_S = 11747 × (1 - 0.549) × 0.221 = 1171 (require no DSP, no VT) P_TrtoSIP = 2287 ÷ 11747 = 0.06

- From the above data, the weighted average penetration factor (PF) is:

  PF = (P_UIPtoUIP × f_1) + (P_UIPtoL × f_2) + (P_LtoL × f_3) + (P_VTtoTr × f_4) + (P_TrtoTr × f_5) + (P_VhtoVs × f_6) + (P_UIPtoVT × f_7) + (P_UIPtoTr × f_8) + (P_LtoVT × f_9) + (P_LtoTr × f_{10}) + (P_VTtoL × f_{11}) + (P_VTtoUIP × f_{12}) + (P_TrtoUIP × f_{13}) + (P_TrtoL × f_{14}) + (P_SIPtoSIP × f_{15}) + (P_SIPtoUIP × f_{16}) + (P_SIPtoL × f_{17}) + (P_SIPtoVT × f_{18}) + (P_SIPtoTr × f_{19}) + (P_VTtoSIP × f_{20}) + (P_TrtoSIP × f_{21}) = (0.03 × 0) + (0.07 × 1.41) + (0.05 × 0.81) + (0.015 × 1.14) + (0.006 × 1.20) + (0.008 × 1.09) + (0.02 × 1.20) + (0.02 × 1.16) + (0.03 × 2.44) + (0.01 × 1.25) + (0.14 × 1.72) + (0.11 × 0.97) + (0.09 × 1.25) + (0.12 × 1.34) + (0.01 × 2.72) + (0.04 × 1.36) + (0.05 × 1.78) + (0.01 × 1.97) + (0.01 × 2.25) + (0.07 × 2.17) + (0.06 × 3.57) = 1.552

- Calculate the System EBC SEBC = (T_{CALL} × (1 + PF + Error_term)) = 19578 × (1 + 1.552 + 0.25) = 54854

Real time calculation with major applications

- ACD agent calls without Symposium $C_{ACD} = [(L_{ACD} + L_{IPACD}) \times 100 \div AHT_{AGT}] = (5610 \times 1650) \times 100 \div 180 = 4033$
- Calculate the impact of other major features and applications.

  You can use Table 67: Feature and application EBC calculation on page 313 to calculate the EBC for your features and applications.

Table 67: Feature and application EBC calculation

Feature or application	EBC calculation forumula	EBC
ACD	$ACDEBC = C_{ACD} \times (1 - \%Symposium) \times f_{ACD}$	= 4033 × 0.1 × 0.13 = 52
Symposium	$SymposiumEBC = \%Symposium \times C_{ACD} \times f_{SYM}$	= 4033 × 0.9 × 5.74 = 20836
CallPilot	$CallPilotEBC = (CP1 + CP2) \times 100 \div AHT_{CP} \times f_{CP}$	= (936 + 624) × 100 ÷ 40 × 1.66 = 6474
Incoming CDR	$IncomingCDR\_EBC = C_{TS} \times f_{CCDR}$	= 11747 × 0.32 = 3759
Outgoing CDR	$OutgoingCDR\_EBC = C_{ST} \times f_{OCDR}$	= 2349 × 0.32 = 752
Tandem CDR	$TandemCDR\_EBC = C_{TT} \times f_{TAN}$	= 587 × 0.44 = 258
Integrated Conference Bridge	MICB_EBC = number of Integrated Conference Bridge ports × $Appl_{CCS} \times 100 \div AHT_{MICB} \times f_{MICB}$	= 16 × 26 × 100 ÷ 1800 × 1.56 = 37
Integrated Recorded Announcer	MIRAN_EBC = number of Integrated Recorded Announcer ports × $Appl_{CCS} \times 100 \div AHT_{MIRAN} \times f_{MIRAN}$	= 12 × 26 × 100 ÷ 90 × 0.63 = 218
Integrated Call Director	MIPCD_EBC = number of Integrated Call Director ports × $Appl_{CCS} \times 100 \div AHT_{MIPCD} \times f_{MIPCD}$	= 12 × 26 × 100 ÷ 60 × 0.63 = 328
Integrated Call Announcer	MIPA_EBC = number of Integrated Call Announcer ports × $Appl_{CCS} \times 100 \div AHT_{MICA} \times f_{MICA}$	= 8 × 26 × 100 ÷ 180 × 0.57 = 66
Hospitality Integrated Voice Service	MIVS_EBC = number of Hospitality Integrated Voice Service ports × $Appl_{CCS} \times 100 \div AHT_{MIVS} \times f_{MIVS}$	= 8 × 26 × 100 ÷ 90 × 0.57 = 132

Feature or application	EBC calculation forumula	EBC
BRI	BRI_EBC = number of BRI users × $Telephone_{CCS}$ × 100 ÷ AHT_{BRI} × f_{BRI}	= 8 × 5 × 100 ÷ 180 × 0.12 = 3
MDECT	MDECT_EBC = L_{DECT} × 100 ÷ WAHT × f_{DECT}	= 1000 × 100 ÷ 128 × 4.25 = 3320
CPND	CPND_EBC = (C_{SS} + C_{TS}) × f_{CPND}	= (4895 + 11747) × 0.2 = 3328
Converged Desktop	CD_EBC = (C_{SS} × 0.1 + C_{TT} + C_{ST} + C_{TS}) × r_{DTP}× f_{DTP}	= (4895 × 0.1 + 587 + 2349 + 11747) × 0.05 × 2.33 = 1768
Features and Applications EBC	FAEBC = ACD_EBC + Symposium_EBC + CallPilot_EBC + InternalCDR_EBC + IncomingCDR_EBC + OutgoingCDR_EBC + TandemCDR_EBC + MICB_EBC + MIRAN_EBC + MIPCD_EBC + MICA_EBC + MIVS_EBC + BRI_EBC + MDECT_EBC + CPND_EBC + CD_EBC + MO_EBC + IPSEC_EBC + MC3100_EBC + MobileX_EBC + ELC_EBC	= 41332

New system real-tme usage

Compare the total system EBC with the CPU rated capacity to determine the processor utilization.

RTU = (SEBC + FAEBC) ÷ BHCC × 100 = (54854 + 41332) ÷ 840 000 × 100 = 12.8 %

In this example, CPU utilization, including application and feature impact, is 12.8 %. This loading indicates that the CPU can handle this configuration with ease and has plenty of spare capacity.

CPU real time conversion for upgrades

If you upgrade a system, in addition to the new load from the above calculation, the CPU utilization data from a current traffic report TFS004 is also required.

Assume you upgrade a system from Communication Server 1000 Release 4 with CP PII to Communication Server 1000 Release 7.5 with CP PIV, when the TFS004 reading is 60%.

From Table 55: Software release degradation factors (SWRC) on page 263 and Table 56: Ratio of existing processor capacity to new processor capacity (CPTU) on page 263:

SWRC = 55 CPTU = 0.32

The calculation to convert the loading is:

CRTU = (60 ÷ 100) × [1 + (55 ÷ 100)] × 0.32 = 0.298

29.8% of the new system CPU (CP PIV) must be reserved to handle calls of the existing site. The expected total CPU utilization is (26.7 + 12.8) = 41.6% .

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:

Number of conference ports = Total_telephones × r_{Con} × 0.4 = 4208 × 0.07 × 0.4 = 207

Number of conference loops = ROUNDUP (conference ports ÷ 30) = 207 ÷ 30 = 7

Number of DSP ports for conference = conference loops × 32) = 7 × 32 = 224

DSP calculation for features and applications

Table 68: DSP ports for features and applications

Feature or application	Number of ports	DSP
Integrated Conference Bridge	16	16
Integrated Recorded Announcements	12 (only 12 listeners)	12
Integrated Call Director	12	12
Integrated Call Assistant	8	8
Hospitality Integrated Voice Service	8	8
BRI	8	16
Agent greeting	4	4
CallPilot	60	60
Total	136	
The general formula for calculating CS 1000E DSP for features and applications is: DSP		

Feature or application	Number of ports	DSP
= number of application ports DSP for BRI = 2× the number of BRI ports		

DSP and Media Card calculations

Total DSP ports = DSP for calls + Conference + Applications/features = 384 + 67 + 73 = 524

Number of 32-port Media Cards required = 524 ÷ 32 = 17

For an 8-port Media Card, number of Media Cards required = 524 ÷ 8 = 66

It is recommended to round up the Media Card calculation to an integer.

See Simplified Media Gateway DSP calculations on page 268 and Table 101: Worksheet B: Detailed DSP and Media Card calculation for Media Gateway on page 406 to determine the card placement and DSP port requirements for each Media Gateway chassis.

Virtual trunk calculation

VT calls (C_{VT}) = C_{T1VT} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} + C_{T2HS} + C_{STSV} + C_{TSVS}= 8184

H.323 VT calls (HC_{VT}) = C_{VT} × V_H = 8184 × 0.67 = 5484

SIP VT calls (SC_{VT}) = C_{VT} × V_S = 8184 × 0.33 = 2701

VT_{CCS} = C_{VT} × WAHT ÷ 100 = 8184 × 128 ÷ 100 = 10476 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 = 347

Number of H.323 Virtual Trunks = 347 × 0.67 = 233

Number of SIP Virtual Trunks = 347 × 0.33 = 115

User input for number of Virtual Trunks is 450. Since this is greater than 347, 450 is the number you use for further resource calculation.

Signaling Server calculation

The following information was obtained from previous calculations or input data:

Signaling Server [type_index = CP PM]

Number of IP Phones in the system (IPL) = 1600 Number of SIP Line Phones in the system (SIPL) = 1200 Number of Virtual Trunks = 450 (H.323 = 300; SIP = 150) HVT = 300 SVT = 150 Calls involving at least one IP Phone (C_{UIP}) = 8267 Calls involving at least one SIP Line Phone (C_{SIP}) = 5277 Calls involving Virtual Trunks (CVT) = GKC0 = 8184

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 or SIP Redirect: Proxy SIP Proxy TCP: Yes SIP Line Alternate (SIPLA) :Yes

1. Signaling Server CP PM Co-resident Call Server and Signaling Server

   Determine if you require a CP PM Co-resident Call Server and Signaling Server.

   The system_type is not CP PM Co-res CS and SS. Follow the Signaling Server algorithm to determine the number of Signaling Servers required.

2. Signaling Server for Personal Directory/Callers List/Redial List

   SSDB = b

   The database share limit for type_index = CP PM is 2000 IP Phones.

3. Network Routing Service calculations

   Dedicated NRS not required, nrs_type_index = type_index = CP PM

   SIP Proxy with TCP protocol is required

   NRS_EP_limit = SIP_Proxy_limit = 1000

   SSNR[nrs_type_index] = larger of:

   {

   a NRE[nrs_type_index] ÷ NRE_EP_limit = 800 ÷ 1000 = 0.8 endpoints

   b NRD[nrs_type_index] ÷ NRD[nrs_type_index] = 1000 ÷ 50 000 = 0.02 dial plan entries

   c NRC[nrs_type_index] ÷ NRC_{HL}[nrs_type_index] = 23486 ÷ 200 000 = 0.15 calls per hour

   d 0 if NRS is not provisioned and there is no branch office = 0

   e If (NRE > NRE_{SL}[nrs_type_index] or NRD > NRD_{SL}[nrs_type_index] or NRC > NRC_{SL}[nrs_type_index]) then 1 else 0 If (800 > 100 or 1000 > 5000 or 28436 > 200 000) evaluates true = 1

   }

SSNR[CP PM] = 1

If SSNR[nrs_type_index] >= 1 or dedicated NRS required

1 >= 1 evaluates true

Then { SSNW = ROUNDUP(SSNR) × NRA (= 2 if true, else 1) = 1 × 2 = 2

SST[nrs_type_index] = SST[nrs_type_index] + SSNW;

SST[CP PM] = 0 + 2 = 2

}

NRC could be hardware or CPU or memory limit, it includes local NRC_0 (calls from main switch calculation) and network VT_{NET} for the NRS:. NRC = NRC_0 + NRC_{NET}

Both VT_{323} and VT_{SIP} from user input must convert to H.323 and SIP calls.

H.323 calls = VT_{323} × CCS × 100 ÷ WAHT = 300 × 28 × 100 ÷ 128 = 6562

SIP calls = VT_{SIP} × CCS × 100 ÷ WAHT = 150 × 28 × 100 ÷ 128 = 3281

Determine the SIP loading factor on the NRS:

Factor = if SIP_mode = Proxy, then 4

NRC_0 = (H323 calls × Factor × SIP calls) = 6562 + 4 × 3281 = 19686

NRC_{NET} = (VT_{NET} × CCS for each VT × 100 ÷ WAHT ÷ 2) = 800 × 28 × 100 ÷ 128 ÷ 2 = 8750

NRC = NRC_0 + NRC_{NET} = 19686 + 8750 = 28436

Formula (c) in SSNR equation = NRC ÷ NRC_{HL}[nrs_type_index] = 28436 ÷ 200 000 = 0.15

4. Terminal Proxy Server calculations

Calculate TPS call rate:

C_{UIP} = $C2_{UIP}$ × 2 + $C1_{UIP}$ + $C2_{SIPUIP}$ + C_{STIV} + C_{STID} + C_{STVI} + C_{STDI}

C_{UIP} = 8267

SSTR[type_index] = larger of:

{

a IPL ÷ IPL $_{SL}$[type_index] = 1600 ÷ 5000 = 0.32

b C_{UIP} ÷ IPC_{HL}[type_index] = 8267 ÷ 40 000 = 0.2

c If IPL > IPL_{DB} then 1 else 0 1600 > 2000 evaluates false = 0

}

SSTR[CP PM] = 0.32

If SSTR[type_index] >= 1 or dedicated TPS required

0.32 >= 1 evaluates false and dedicated not required

Else { If SSTR[type_index] > 0

0.32 > 0 evaluates true

Then { TPS_Co-res = true; NumOfCo-res = NumOfCo-res + 1

= NumOfCo-res = 0 + 1 = 1

}

}

5. H.323 Gateway calculations

HC_{VT} = (HVT_{CCS} × 100) ÷ WAHT

= (8400 × 100) ÷ 128

= 6562

SSHR[type_index] = larger of:

{

a HVT[type_index] ÷ HVT_{SL}[type_index] = 300 ÷ 1200 = 0.25

b HC_{VT}[type_index] ÷ $HVTC_{HL}$[type_index] = 6562 ÷ 18 000 = 0.36

c If HVT > $VTRK_{SL}$[type_index] then 1 else 0 300 > 800 evaluates false = 0

}

SSHR[CP PM] = 0.36

If SSHR[type_index] >= 1 or dedicated H323 GW required

0.36 >= 1 evaluates false and dedicated not required

Else { If SSHR[type_index] > 0

0.36 > 0 evaluates true

Then { H323_Co-res = true; NumOfCo-res = NumOfCo-res + 1

= NumOfCo-res = 1+ 1 = 2

}

}

6. SIP Gateway calculations

SC_{VT} = (SVT_{CCS} × 100) ÷ WAHT

= (4200 × 100) ÷ 128

= 3500

SSSR[type_index] = larger of:

{

a $SVT[type\_index] \div SVT_{SL}[type\_index] = 150 \div 1800 = 0.08$

b $SC_{VT}[type\_index] \div SVTC_{HL}[type\_index] = 3500 \div 27\,000 = 0.13$

c SVT > $VRTK_{SL}$[type_index] then 1 else 0 150 > 800 evaluates false = 0

}

SSSR[CP PM] = 0.13

If SSSR[type_index] >= 1 or dedicated SIP GW required

0.13 >= 1 evaluates false and dedicated not required

Else { If SSSR[type_index] > 0

Then { SIP_Co-res = true; NumOfCo-res = NumOfCo-res + 1

= NumOfCo-res = 2+ 1 = 3

}

}

7. SIP CTI/TR87 calculations

   No SIP CTI/TR87 specified.

8. Signaling Server Co-resident calculations

Determine if any Signaling Server applications can co-reside on one Signaling Server

Case of NumOfCo-res:

NumOfCo-res = 3, evaluate the portion if more than one Signaling Server application is Co-resident

Determine if Co-resident VRTK exceeds limit

If H323_Co-res = true and SIP_Co-res = true and (HVT + SVT > VT_{SL}[type_index]) H323_Co-res = true and SIP_Co-res = true and (300 + 150) > 800) This evaluates false;

If NumOfCo-res = 0, then exit NumOfCo-res = 3, evaluates false

Else If NumOfCo-res = 1, then do, one Signaling Server application - assign one Signaling Server NumOfCo-res = 3, evaluates false

Else { CallRate = 0;

If NRS_Co-res = true, then CallRate = CallRate + NRC; NRS_Co-res evaluates false

If TPS_Co-res = true, then CallRate = CallRate + C_{UIP}; TPS_Co-res evaluates true, CallRate = 8267;

If H323_Co-res = true, then CallRate = CallRate + HC_{VT}; H323_Co-res evaluates true, CallRate = 8267 + 6562 = 14829;

If SIP_Co-res = true, then CallRate = CallRate + SC_{VT}; SIP_Co-res evaluates true, CallRate = 14829 + 3500 = 18329;

If CallRate <= Co-resCR[type_index] 18329 <= 20000 Co-res CallRate limit met

Then { SST[type_index] = SST[type_index] +1; SST[CP PM] = 1 + 1 = 2;

Since one of the following evaluates true; SST[CP PM] = 2 + 1 = 3

If (NRS_Co-res = true and NRA = true) or (TPS_Co-res = true and TPSA = true) or (H323_Co-res = true and GWA = true) or (SIP_Co-res = true and GSA = true) or (TR87_Co-res = true and TR87A = true)

Then SST[type_index] = SST[type_index] + 1;

}

}

}

9. SIP Line Gateway calculations

SIPL = SIPN + SIP3, where SIPL = total number of SIP Phones SIPL = 1200

Calculate total number of SIP Line calls

$C_{SIP} = (2 \times C2_{SIP}) + C1_{SIP} + C2_{SIPUIP} + C_{STSV} + C_{STSD} + C_{STVS} + C_{STDS}$

$C_{SIP} = 5277$

SSSLGR[type_index] = larger of:

{

a $SIPL \div SIPL_{SL}$[type_index] = 1200 ÷ 1000 = 1.2

b $C_{SIP} \div SIPLC_{HL}$[type_index] = 5277 ÷ 10000 = 0.53

}

SSSLGR[CP PM] = 1.2

Round up SSLGR before you proceed with the calculations

SSSLGW[type_index] = ROUNDUP(SSSLGR[type_index] × SIPLA[type_index] (= 2 if true, else 1)

SIPLA = if redundant SIPL is needed

SSSLGW[CP PM] = 2 × 2 = 4

SST[type_index] = SST[type_index] + SSSLGW; SST[CP PM] = 3 + 4 = 7

10. SIP DECT calculations

    No SIP DECT Phones required.

11. Total Signaling Server requirement

    The total number of Signaling Servers provisioned:

    SST[type_index] = 6, and If nrs_type_index and type_index are not the same, then SST[nrs_type_index], else 0 nrs_type_index [CP PM] = type_index [CP PM] evaluates 0, and Signaling Servers for SIP DECT = SS_DECT = 0

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

Chapter 15: Application engineering

Contents

This chapter contains the following topics:

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 Avaya publications.

For more information about voice networks, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Access Restrictions

The Access Restrictions feature, also known as the port blocking facility is a VxWorks-based firewall designed to prevent port-based attacks on the CP PIV, MGC, and MC32S running VxWorks software. Access Restrictions use port blocking rules for accepting or rejecting packets to open ports. The port blocking rules are preconfigured during installation and distribute from the Call Server to the MGC and MC32S. You can customize the port blocking rules post installation with Overlay 117 or EM.

Adding port blocking rules increases CPU utilization. Avaya recommends you to maintain minimal port blocking rules to minimize the CPU performance impact. Access Restrictions provide a minimum but essential firewall to secure the VxWorks platforms. If you require a full firewall, Avaya recommends the use of a dedicated third-party hardware firewall.

CPU utilization depends on the type and amount of rules configured. Table 69: CP PIV packet throughput drop at 10 percent CPU utilization on page 324 provides an example of the CP PIV performance drop with increasing rule depth.

Table 69: CP PIV packet throughput drop at 10 percent CPU utilization

Rule Depth	60 byte packet, reject at end. (packets/ second)	60 byte packet, accept at rule. (packets/ second)	60 byte packet, accept at rule. CPU utilization drop against no firewall.	60 byte packet, accept at rule. CPU utilization drop against accept all default rule.
No firewall	66 500	66 500	0	n/a
0	64 000	57 000	14.3%	0
1	57 000	51 000	23.3%	10.5%
4	52 000	48 000	27.8%	15.8%
8	43 000	41 000	38.3%	28.1%
16	35 000	33 000	50.4%	42.1%
32	24 750	25 000	62.4%	56.1%
64	13 250	13 500	79.7%	76.3%
128	7 500	7 500	88.7%	86.8%

Converged Desktop

The Converged Desktop is a TDM or IP Phone configured to access Avaya Multimedia Communication Server 5100 (Avaya MCS 5100) multimedia applications through a Session Initiation Protocol (SIP) Virtual Trunk.

Maximum number of Converged Desktop users

In a pure IP system, the Avaya Communication Server 1000E (Avaya CS 1000E) can support up to 10 000 Converged Desktop users. However, for a new installation, Avaya 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 nonSIP 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 70: SIP port and PCA requirements for Converged Desktop (with P.01 GoS) on page 327 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 70: SIP port and PCA requirements for Converged Desktop (with P.01 GoS) on page 327 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 70: 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
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

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
	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
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

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
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
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
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

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
	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	10 000.0
	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
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	12 500.0
	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	15 000.0
	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

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
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	17 500.0
	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	10 100.0	11 000.0	20 000.0
	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
4500	SIP CCS	2250.0	3262.5	4275.0	5287.5	6300.0	7312.5	8325.0	9337.5	10 350	11 362.5	12 375.0	22 500.0
	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	10 375	11 500	12 625	13 750	25 000
	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	11 100	12 450	13 800	15 150	16 500	30 000
	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	11 375	12 950	14 525	16 100	17 675	19 250	35 000

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
	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
8000	SIP CCS	4000	5800	7600	9400	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	4500	6525	8550	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	5000	7250	9500	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.													

Exchange 2007 Unified Messaging SIP trunk provisioning

For information about Exchange 2007 Unified Messaging SIP trunk provisioning, see *Avaya Communications Server 1000 with Microsoft Exchange Server 2007 Unified Messaging Fundamentals, NN43001-122*. Refer to the SIP provisioning guidelines section.

Microsoft Office Communications Server users

The Avaya Converged Office feature combines the business-grade telephony of the Avaya CS 1000 with the real time multimedia communication and the remote call control provided by Microsoft® Office Communications Server (OCS) 2007. Avaya Converged Office comprises the following components:

- Remote Call Control (RCC) with Session Initiation Protocol (SIP) Computer Telephone Integration (CTI) TR/87 provides full Microsoft® Office telephony integration to control business-grade telephones 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 Office Communicator (OC) 2007 clients.

Trunking

To handle the traffic between the Communication Server 1000 and the Office Communications Server 2007, you must configure sufficient SIP trunks and Universal Extensions (UEXT). 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 × 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).

Telephony services (TLSV) has replaced Personal Call Assistant (PCA). TLSV extends the call over a SIP trunk to the OCS client from the Communication Server 1000 system.

Calculating SIP access port and TLSV requirements

the following table defines the inputs used to calculate SIP access ports and TLSV requirements.

Table 71: Inputs

Input	Description

TN_MO_Users	Total number of Office Communicator users that utilize the SIP Access Ports for voice services.
UEXT_MO_Users	Number of Office Communicator users that use Universal Extension (UEXT) with Telephony Services (TLSV) subtype. The UEXTs you require are additional to the number of UEXTs indicated in the Enterprise Configurator (EC) tool software.
P_UEXT_SIP	Percentage of UEXT calls that use the soft client to answer a call.

Calculations

Use the following formulas to calculate traffic requirements:

Traffic for UEXTs = (UEXT_MO_Users) × (CCS per user) × (1 - P_UEXT_SIP) × 10%

Traffic for SIP ports = (TN_MO_Users - UEXT_MO_Users) × (CCS per user) + (UEXT_MO_Users × P_UEXT_SIP) × (CCS per user)

Total SIP Traffic = (Traffic for UEXTs) + (Traffic for SIP ports)

Number of MO SIP ports = Poisson (Total SIP Traffic) at P.01 Grade of Service

MO = Microsoft® Office Communicator

Table 72: Traffic figures on page 334 shows traffic in CCS and number of ports calculated based on Poisson formula at P.01 Grade of Service.

Table 72: 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	9
150	4.14	10
175	4.86	12
200	5.56	13
225	6.25	14
250	6.94	15
275	7.64	16
300	8.33	17
325	9.03	18
350	9.72	19
375	10.42	19
400	11.11	20
425	11.81	21
450	12.5	22
475	13.19	23
500	13.89	24
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

Basic Client Configuration

The Basic Client Configuration (BCC) can program the new TLSV subtype for UEXT TNs. All UEXTs associated with OCS 2007 require the TLSV subtype.

LD 11 supports the administration of telephones. BCC uses REQ commands, such as NEW, CHG, and OUT. In LD 20, BCC uses the PRT command to retrieve phones from the Call Server.

Port use

The Communication Server 1000 uses the following ports for TCP and TLS:

- 5060: TCP
- 5061: TLS

The dynamic port range Office Communicator uses for SIP and RTP is 1024 - 65535. You can restrict the port range with group policy settings. Port ranges must not overlap. For more information, see the help and support page on the Microsoft Web site at http://www.microsoft.com.

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 on page 338.

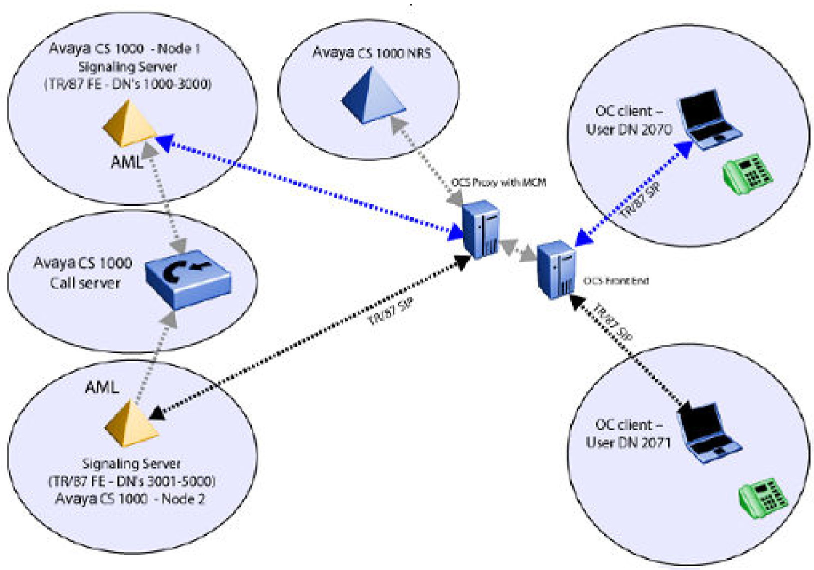

Figure 57: Capacity example

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. One Signaling Server can support up to 1800 SIP trunks, therefore you require two Mediation servers for each Signaling Server to correctly deploy OCS 2007. To increase the system capacity, associate a pool of Mediation servers with each Call Server. The Multimedia Convergence Manager (MCM) routes inbound calls from the Signaling Server to the appropriate Mediation server within the Mediation server pool. The CP PIV and CP PM Call Server can support up to 13 000 users.

For more information about Converged Office features and engineering, see *Avaya Converged Office Fundamentals - Microsoft Office Communications Server 2007, NN43001-121*.

Mobile Extension engineering

The following sections detail the engineering related to your use of Mobile Extensions, Primary Rate Interface, Digital Signal Processor, and Digitone Receiver resources for Mobile Extension.

Mobile Extension

You can configure a mobile user with a Mobile Extension (MOBX), providing a logical connection to the users mobile phone. Each mobile user requires a configured MOBX.

There is a limit of 4000 Mobile Extensions per customer.

MOBX Digital Signal Processor engineering

You require a Digital Signal Processor (DSP) for each Primary Rate Interface (PRI) trunk allocated for MOBX. The DSP resources are used for dual tone multi frequency (DTMF) detection. DSP resources are chassis and cabinet specific on a CS 1000E system. You must have a one to one relationship with PRI trunks in each chassis or cabinet. See MOBX Primary Rate Interface engineering on page 340 to calculate the DSPs required.

MOBX Digitone Receiver engineering

You require additional Digitone Receivers (DTR) for the MOBX beyond the required DTRs for telephones or Digitone trunks. Invoking the Mobile Feature Activation Code (MFAC) disables the DSP DTMF detector and regular DTR resources are required for the Flexible Feature Code handling. The MOBX DSP resource is released when the call is released. You can calculate the number of DTRs required using the following two formulas.

MOBX DTR engineering example

NC = (100 x T) / CHT

Where NC = number of calls

T = traffic in CCS

CHT = 150 (average call hold time in seconds)

DTR CCS = (HT x NC) / 100

Where HT = 6 (DTR average hold time in seconds)

Use your DTR CCS calculated call value and see Table 113: Digitone receiver load capacity 6 to 15 second holding time on page 422 to find the number of DTRs required under the 6 second column. DTR resources are chassis and cabinet specific on a CS 1000E system. Avaya recommends you configure all 16 DTRs for each chassis or cabinet with MOBX.

MOBX Primary Rate Interface engineering

You can calculate the number of PRI trunks required for MOBX by using the call rate of MOBX users. (See Table 107: Trunk traffic Poisson 1 percent blocking on page 415).

MOBX PRI engineering example

1000 MOBX users

5 CCS busy hour call attempts (BHCA)

Therefore, 1000 x 5 = 5000 CCS

The table indicates 5000 CCS requires 168 trunks and 168 DSPs, a one to one relationship between DSP and DTR trunks.

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.

Multiple D-channels

Avaya does not recommend you to split the Primary and Backup D-channels of the same ISDN Trunk Group across multiple GR/CR CS 1000E Media or PRI Gateways. While this configuration insures D-channel redundancy during some Primary D-Channel failure situations, states could exist where both D-channels register to different Call Servers and simultaneously activate creating a conflict in the Central Office. This conflict can affect service and can lead to a complete ISDN Trunk Group outage in most service provider Central Offices.

If your service provider supports ISDN Trunk Group hunting, Avaya recommends you to maintain multiple ISDN Trunk Groups with each ISDN service provider. Configure each Trunk Group with its own Primary and Backup D-channels on PRI circuits in each Media Gateway. This solution offers resilient configuration in larger systems distributed geographically and operates well even if your service provider is unable to support a D-channel for each ISDN PRI circuit.

Avaya can provide VoIP Session Border Controllers as an alternative to large scale ISDN Trunking facilities. This solution offers improved flexibility in deployment and resiliency performance. For more information, see www.avaya.com/support.

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 *Avaya 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 73: ISL link capacities on page 344). 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 on page 342.

Table 73: 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 73: ISL link capacities on page 344, the link is the limiting constraint. The potential peak traffic problems described in Peak analysis on page 343 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.

Avaya 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 Avaya 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)

For Erlang B values, see Trunk traffic Erlang B with P.01 Grade-of-Service on page 413.

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) 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 74: Link capacities for CDR application (outgoing) on page 350). 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 74: 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. For more information about real time engineering of the system, see *Avaya Traffic Measurement Formats and Outputs Reference, NN43001-750* . 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 75: D-channel handler engineering worksheet on page 351 is a high-level worksheet for analysis of D-channel handling capacity. See Table 81: Real time requirements for D-channel applications on page 356 through Table 79: Peak buffer requirements for SDI applications on page 353 for the values to use in the worksheet.

Table 75: 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

Avaya 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 75: D-channel handler engineering worksheet on page 351 through Table 79: Peak buffer requirements for SDI applications on page 353, 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 75: D-channel handler engineering worksheet on page 351, and the appropriate Peak Buffer Usage values should be inserted in the corresponding Peak Buffer Usage columns of Table 75: D-channel handler engineering worksheet on page 351.

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 81: Real time requirements for D-channel applications on page 356.

Table 76: 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	_____
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+," see D-channel on page 342					

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 on page 354 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 *Avaya Traffic Measurement Formats and Outputs Reference, NN43001-750* for details.

Table 77: 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

DCH	Outgoing	Incoming
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 81: Real time requirements for D-channel applications on page 356.

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 Avaya CallPilot agents in the agent total. This compensates for the assumption of simple calls.

Table 78: 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 about 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 79: Peak buffer requirements for SDI applications

SDI	Outgoing	Incoming	Minimum baud rate
CDR	• 30 if baud rate is less than recommended in Table 74: Link capacities for CDR application (outgoing) on page 350	1	(msgs/hr × 10 bits/msg) ÷ (3600 sec/hr) = _____

SDI	Outgoing	Incoming	Minimum baud rate
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 on page 355. 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 80: NACD Message Model on page 355).

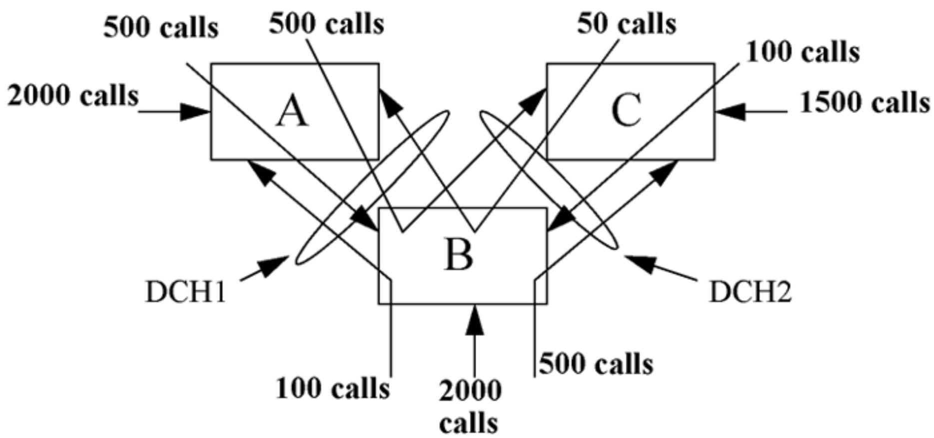

Figure 58: NACD network

Table 80: 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	
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 81: Real time requirements for D-channel applications on page 356).

Table 81: 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 nonNACD calls are carried, Node B carries 3750 calls/hour.

Table 82: 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=3750	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 83: 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.

Avaya CallPilot engineering

For information about Avaya CallPilot engineering, see *CallPilot Planning and Engineering, 555-7171-101*. 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 nonvoice application reduces the number of channels available for voice traffic.

For an IP source to access Avaya CallPilot, the codec must be set for G.711. Since a nonstandard 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. For more information, see *CallPilot Planning and Engineering, 555-7101-101* .

Once the CCS for each type of media is calculated, sum up the total and refer to capacity tables in the publication for the MPU requirement based on the offered CCS traffic.

For nonblocking 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 Avaya 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 autoterminating 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 document 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 nonblocking 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 an Avaya 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.

For more information, see Service loops and circuits on page 220.

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 Avaya 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 nonblocking access, provide one DSP port for each ACD agent configured.

For the detailed calculations, see Resource calculations on page 249.

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 Avaya 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 autonegotiate, is not a bottleneck in a Symposium/ CallPilot configuration. However, observe the following engineering guidelines to avoid performance problems. For more information, see *Avaya 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.

Survivable and Distributed Media Gateway ELAN traffic estimation

When using a Survivable or Distributed Media Gateway in environments with bandwidths less than 20Mbit/s, Avaya recommends to estimate the ELAN traffic bandwidth for each Media Gateway. This information is required to properly plan and engineer the data network needs of the gateway.

Failure to estimate and engineer the data network for the ELAN traffic for Survivable or Distributed Media Gateways may result in unpredictable behavior of the Media Gateway under load conditions. Avaya also recommends routinely monitoring the ELAN traffic to determine the actual bandwidth needs for the Media Gateways. Any configuration change to Media

Gateways requires a recalculation of the ELAN traffic estimation to ensure proper data networking.

You can estimate the ELAN traffic by estimating the load on each card in the Media Gateway in an estimation table.

Table 88: Estimated Elan Traffic by IPE card (by card function) on page 365 lists estimated bandwidth by IPE card type. Use this table to select the estimated bandwidth requirements by the type of card placed in a media gateway.

The method of determining the ELAN traffic for a media gateway is as follows:

- Select a table that represents the type of media gateway that is being estimated (MG 1000, MG 1010, MG XPEC, MG PRI Gateway).
- For each IPE card placed in the gateway, determine the expected traffic load in CCS for the card.
- Use Table 88: Estimated Elan Traffic by IPE card (by card function) on page 365 to select a CCS rate that is closest to the expected rate (normal or high), then copy the appropriate bandwidth number, including the idle traffic bandwidth, for each card. For non-blocking cards, you must select the high traffic rate.
- Sum both the Idle bandwidth and Traffic bandwidth columns.
- The total bandwidth for the media gateway is the sum of the idle bandwidth and traffic bandwidth values.

If a card is not found in Table 88: Estimated Elan Traffic by IPE card (by card function) on page 365 , use the Unknown IPE Card (UIC) with a high traffic load.

If a card function has more than one entry, such as the UTC, this is due to a market specific version of the card that is using a different “normal” traffic load. If the version of the card you are using does not match that market specific card, use the “any version” entry.

Table 84: Example 1 Media Gateway ELAN Estimation on page 361 shows an example ELAN traffic estimation table for an MG 1000 media gateway.

The calculation is the same for any media gateway. Add the traffic for the number of cards (of the correct type) for the type of gateway in use, including Idle traffic and Traffic Load.

Table 85: Example 2 Media Gateway ELAN Estimation on page 362, Table 86: Example 3 PRI Media Gateway ELAN Estimation on page 363, and Table 87: Example 4 MG XPEC ELAN Estimation on page 364 show ELAN bandwidth estimations for other types of media gateways.

Table 84: Example 1 Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
10	DLC	Normal	0	20800
9	DLC	Normal	0	20800

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
8	DLC	Normal	0	20800
7	DLC (for ACD Agents)	High	0	36400
6				
5				
4	TMDI (1.5 Mb)	High	14400	100000
3	DTI (1.5 Mb)	High	14400	100000
2	MW ALC	Normal	0	240000
1	PRI (1.5 Mb)	High	14400	100000
0	CPMG128 (128 DSP)	Normal	6000	(see Note on page 362)
		Totals	49240	422800
		Chassis total	472,040	
		With IPsec ON	613,652	

Based on the card placement above, the estimated ELAN requirement for the chassis is as follows:

49,240 + 422,800 = 472,040 Bits/sec

With IPSec, there is an estimated 30% increase in traffic overhead. ELAN traffic estimation for the same chassis with IPSec is 613,652 Bits/sec.

Note:

All Media Gateway traffic is directed to a CS 1000 Call server card; therefore, the active Call Server will see the sum of ELAN traffic from all Media Gateways that are registered to that Call Sever. The Call Server also sees all of the ELAN Traffic from all registered Signaling Servers.

In normal operations, the Call Server is assumed to be on the main campus with a full duplex 100BT connection. If the Call Server card is for survivability and is not the active Call Server, then there is no ELAN traffic for this card.

Table 85: Example 2 Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
10	DLC	Normal	0	20800
9	DLC	Normal	0	20800

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/ sec)	Traffic Load (Bits/ sec)
8	DLC	Normal	0	20800
7	DLC (for ACD Agents)	High	0	36400
6	UTC	Normal	0	12000
5	ALC	Normal	0	24000
4	DTI (2.0 Mb)	High	14400	125000
3	MIRAN	High	0	36400
2	MW ALC	Normal	0	24000
1	PRI (2.0 Mb)	High	14400	125000
0	MGC (128 DSP)	Normal	6000	(see Note on page 362)
23	CP DC — Call Server	Normal	(see Note on page 362)	(see Note on page 362)
22				
		Totals	34800	445,200
		Chassis total	480,000	
		With IPsec ON	624,000	

Based on the card placement above, the estimated ELAN requirement for the chassis is as follows:

34,800 + 445,200 = 480,000 Bits/Sec

Table 86: Example 3 PRI Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/ sec)	Traffic Load (Bits/ sec)
2	PRI Gwy Exp. (2.0 Mb)	High	57600	500000
1	PRI Gwy Base (2.0 Mb)	High	57600	500000
0	CPMG (256 DSP)	Normal	6000	
		Totals	121200	100000
		Chassis total	1,121,200	
		With IPsec ON	1,457,560	

Based on the card placement above, the estimated ELAN requirement for the chassis is as follows:

121,200 + 1,000,000 = 1,121,200 Bits/Sec

Table 87: Example 4 MG XPEC ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/ sec)	Traffic Load (Bits/ sec)
15	DID	Normal	0	12000
14	EM Trunk	High	0	24000
13	DLC	Normal	0	20800
12	DLC (for ACD agents)	High	0	36400
11	DLC	Normal	0	20800
10	DLC (for ACD agents)	High	0	36400
9	DLC	Normal	0	20800
8	DLC	Normal	0	20800
	MGXPEC		12000	0
7	DLC	Normal	0	20800
6	DLC (for ACD agents)	High	0	36400
5	UTC	Normal	0	12000
4	ALC	Normal	0	24000
3	MW ALC	Normal	0	24000
2	MIRAN	High	0	36400
1	MW ALC	Normal	0	24000
0	MW ALC	Normal	0	24000
		Totals	12000	393,600
		Chassis total	405,600	
		With IPsec ON	527,280	

Based on the card placement above, the estimated ELAN requirement for the chassis is as follows:

12,000 + 393,600 = 405,600 Bits/Sec

Table 88: Estimated Elan Traffic by IPE card (by card function)

Short Description	Example PEC	Idle Traffic (Bits/ sec)	Normal Traffic (Bits/ sec)	Normal Load (CCS)	High Traffic (Bits/ sec)	Max. Load (CCS)	Default to High
2.0Mb DTI	any version	14400	57500	144	125000	949	Yes
2.0Mb PRI	any version	14400	57500	144	125000	949	Yes
48port DLC	NTDK16	0	62400	240	109200	1584	
Agent Greeting	NTVQ09AB	0	20800	80	36400	528	
ALC	any version	0	24000	80	42000	528	
Call Pilot Mgate	NTRB18	0	20800	80	36400	528	
CallPltIPE	NTUB01	0	20800	80	36400	528	
Call Server (CP MG) (see Note on page 362)	any version	6000	0	0	0		
Call Server (CP PM, CP DC, COTS2, COTS3) (see Note on page 362)	any version	0	0	0		0	
DID	any version	0	12000	40	24000	264	
DID/DDI	any version	0	12000	40	24000	264	
DLC (Digital Line Card)	any version	0	20800	80	36400	528	
DMC4	NTCW00	0	20800	80	36400	528	
DMC4E	NTCW01	0	20800	80	36400	528	
DMC8	NTCW00B	0	20800	80	36400	528	
DMC8E	NTCW01B	0	20800	80	36400	528	
DXUT	NT5D39	0	20800	80	36400	528	
EM Trunk	any version	0	12000	40	24000	264	
FCOT	any version	0	12000	40	24000	264	
ICA	NT5G01	0	20800	80	36400	528	
ICB 4	NT5D51BC	0	20800	80	36400	528	

Short Description	Example PEC	Idle Traffic (Bits/ sec)	Normal Traffic (Bits/ sec)	Normal Load (CCS)	High Traffic (Bits/ sec)	Max. Load (CCS)	Default to High
ICD	NT5G71	0	20800	80	36400	528	
ITG	NTVQ55	0	20800	80	36400	528	
ITG	NTZC13	0	20800	80	36400	528	
Line side E1	NT5D34	14400	57500	144	125000	949	
Line side T1	NT5D14	14400	46000	115	100000	759	
MG XPEC	any version	12000	0	0	0	0	
Mess. Waiting ALC	any version	0	24000	80	42000	528	
MGC (includes DSP)	any version	6000	0	0	0	0	
MICB	any version	0	20800	80	36400	528	
MIRAN	any version	0	20800	80	36400	528	Yes
MISP 11C	NTBK22	14400	46000	115	100000	759	Yes
MIVS	NT5G15	0	20800	80	36400	528	
OPX	NTRA06	0	24000	80	42000	528	
Inc. 3 wire Trk	NT5K60	0	12000	40	24000	264	
Out. 3 wire Trk	NT5K61	0	12000	40	24000	264	
PRI/DTI (1.5 Mb)	any version	14400	46000	115	100000	759	Yes
PRI Gateway (base 4 spans) (1.5 Mb) (2.0 Mb)	any version	 57600 57600	 184000 230000	 460 576	 400000 500000	 3036 3796	Yes
PRI Gateway (exp. 4 spans) (1.5 Mb) (2.0 Mb)	any version	 57600 57600	 184000 230000	 460 576	 400000 500000	 3036 3796	Yes

Short Description	Example PEC	Idle Traffic (Bits/ sec)	Normal Traffic (Bits/ sec)	Normal Load (CCS)	High Traffic (Bits/ sec)	Max. Load (CCS)	Default to High
Remote Agent Observ.	NTVQ01BA	0	20800	80	36400	528	
RLC (single port card)	NTDR68	0	5200	5	5200	33	
RLC III	NTDR71	0	48000	160	84000	1056	
SDI/DCH	NTAK02	0	20800	80	36400	528	Yes
SILC	NT6D70	14400	15333	38	33333	253	
SMC	NTVQ01	0	20800	80	36400	528	
TDS/DTR	NTAK03	0	20800	80	36400	528	Yes
TMDI (1.5Mb DTI/ PRI)	any version	14400	46000	115	100000	759	Yes
UILC	NT6D71	14400	15333	38	33333	253	
Unknown IPE Card	—	0	20800	80	36400	528	Yes
UTC (Univ. Trunk Card)	NT5D26	0	20800	80	36400	528	
UTC (Univ. Trunk Card)	NT5D31	0	20800	80	36400	528	
UTC (Univ. Trunk Card)	NT5K07	0	24000	80	42000	528	
UTC (Univ. Trunk Card)	any version	0	12000	40	24000	264	
XCMC	any version	0	20800	80	36400	528	Yes
XDDI	NTAG04	0	12000	40	24000	264	
XFCOT	NT5D29	0	24000	80	42000	528	
XFCOT	any version	0	12000	40	24000	264	
XMFC/MFE	NT5K21	0	20800	80	36400	528	Yes
XMFR	NTAG36	0	12000	40	24000	264	Yes
XOPS	NT1R20	0	24000	80	42000	528	
XUT Trunk	any version	0	12000	40	24000	264	

- ELAN Traffic is additional traffic from the ELAN ports on the MGC that must be included in the overall data network bandwidth estimation. This traffic is specifically between the

MGC and the controlling Communication Server. The MGC may re-home to 3 Communication Servers.

- Idle Traffic (Bits/sec) is the minimum ELAN traffic that occurs under no load conditions. This traffic is always present and must be added to the total ELAN traffic estimation.
- Normal Traffic (Bits/sec) is the normal ELAN traffic during 20% card load or 5 CCS for each port. The 20% card load is 20% of ports having simultaneous state changes.
- Normal Load CCS is the normal card load measured in CCS. For example, the Digital Line Card (DLC). Digital Line Card under normal CCS is 80 CCS (16x5CCS = 80 CCS).
- High Traffic (Bits/sec) is the maximum ELAN traffic during 40% card load at 33 CCS for each active port. The 40% card load is defined as 40% of the ports having simultaneous state changes. All ports can be busy. An active port without state changes has no ELAN messaging above the idle state.
- Maximum Load CCS is the maximum card load measured in CCS. For example, the Digital Line Card under maximum CCS is 528 CCS (16x33CCS = 528 CCS).

Use this table only for estimating the Media Gateway ELAN traffic. Do not use these table values for other calculations.

Additional network parameters, such as TLAN traffic, packet loss, and round trip delay, are also required for proper data network planning and engineering. See *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260* for additional details on Distributed Media Gateways.

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 information about how the Campus Redundancy feature works, see *Avaya 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. Avaya strongly recommends 802.1p Level 7 (Network Control and OAM).
- When using third-party vendor network equipment that has not been validated by Avaya, 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 autonegotiate the link speed and duplex. Therefore, the network equipment that the CP PIV is attached must also use autonegotiate. Verify that both the CP PIV and the network equipment speed and duplex are a match.
- Avaya 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 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 autonegotiate and the other is hard configured.

The autonegotiate side adapts only to the speed setting of the fixed side. For duplex operations, the autonegotiate 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

Important:

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. Avaya strongly recommends allocation of a network IP address within a customer address space if the network is not dark fiber driven by Ethernet Routing Switch 2550T.

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 nonblocking 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 89: CMOD Unit Capacity on page 372 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 89: 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
28	1855-1949	60	5133-5239
29	1950-2046	61	5240-5347
30	2047-2142	62	5348-5455

CMOD Unit	CLASS Telephone	CMOD Unit	CLASS Telephone
31	2143-2240	63	5456-5563
32	2241-2338	64	5564-5671

Guidelines for nonCall Center applications

In a noncall 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 89: CMOD Unit Capacity on page 372).

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 89: CMOD Unit Capacity on page 372).

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 on page 197 and Memory engineering on page 209.

Media Application Server (MAS)

MAS servers are deployed in clusters, with a maximum cluster size of 7 servers.

Within a cluster, one server is designated at the Primary MAS server, which handles the licensing and load sharing for all of the servers within that cluster. Optionally, a Secondary MAS server can be designated, which will take over the licensing and load sharing function if the Primary server fails. Both of these servers participate in handling the load (conference, tone, RAN, Music).

Note:

If both the Primary and Secondary servers in a cluster fail, the remaining servers in the cluster no longer function, as they cannot obtain a license.

MAS license keys are not shared between MAS clusters.

The MAS license server does not know the capacity limitations of the MAS servers in the cluster. Therefore, all MAS servers within the cluster **MUST** be of the same type, or at a minimum, have the same capacity rating.

When a redundant MAS server is required, an additional MAS server is added to each cluster. This is done to ensure the MAS servers can handle the load of a cluster in the event that one of the servers fails or goes offline. The load capacity of this redundant server is not factored into the calculation of load capacity of the cluster as a whole.

There is a calculation used to determine the number of MAS servers required:MAS Calculation on page 301. A number of the variable definitions and calculated values from the MAS calculation are required to determine how to divide the Total_MAS_Session_Licenses between the different MAS clusters.

It should be noted that the MAS Codec selected and the use of Media Security impact the load capacity of a MAS server. The MAS server engineering assumes the same codec is used for all sessions on all MAS servers.

The number of clusters and the size of the license key for each cluster must be calculated.

MAS License/Keycode Calculation:

The total number of MAS sessions equates to the total session licenses required in all MAS keycodes.

Total_MAS_Session_Licenses = MSC_sessions+XMSC_sessions

Each cluster of MAS servers must have its own keycode. Therefore the total number of MAS keycodes is equal to the number of MAS clusters.

First, calculate the number of MAS clusters required for the number of servers that will be deployed (MAS_Clusters).

If (MASA[MASA_type_index] = true then

MAS_Clusters = round_up(SSMASR / (MAS_Cluster_size -1))

Else

MAS_Clusters = roundup(SSMASR / MAS_Cluster_size)

The MAS licenses need to be divided up between the clusters and the number of servers in each cluster must be taken into consideration.

Second, calculate the number of servers in each Cluster:

Where

Servers_in_cluster_one_and_full is defined as the number of servers in the first cluster and the number of servers in each full cluster. Therefore when there is only one cluster this number can be less than the cluster size.

Servers_in_cluster_N is defined as the number of servers in the last cluster. When there is only one cluster this value will be zero. This value will often be smaller than the cluster size.

```
If (MASA[MASA_type_index] = true then % have redundancy
{
    Servers_in_cluster_one_and_full = MAS_Cluster_size -1
    If MAS_Clusters > 1 then
        Servers_in_cluster_N = roundup(SSMASR-
        ((MAS_Clusters-1)*MAS_cluster_size-1))
    Else % only one cluster
        Servers_in_cluster_N =0
}
Else % no redundancy
{
    Servers_in_cluster_one_and_full = MAS_Cluster_size
    If MAS_Clusters > 1
    then
        Servers_in_cluster_N = roundup(SSMASR-
        ((MAS_Clusters-1)*MAS_cluster_size))
    Else % only one cluster
        Servers_in_cluster_N = 0
```

}

Now that the number clusters is known and the number of servers per cluster is known, the size of the keycode per cluster can be calculated.

Note that the cluster N-1 may not be a full cluster and therefore will have a different number of licenses than a full cluster.

If MAS_Clusters > 1 then

{

Numb_Cluster_one_keycodes = MAS_Clusters – 1
Numb_ClusterN_keycodes = 1
If MASA = true % have redundancy

{

Cluster_one_keycode_Sessions =

ROUNDUP(Total_MAS_Sessions_Licenses/SSMASR*(MAS_cluster_size-1)

}

Else % no redundancy MAS servers

{

Cluster_one_keycode_Sessions =

ROUNDUP(Total_MAS_Sessions_Licenses/SSMASR*(MAS_cluster_size)

} % end redundancy check

% there is more than one cluster, so determine keycode sessions for clusterN

ClusterN_keycode_Sessions = Total_MAS_Session_Licenses –

(Cluster_one_keycode_Sessions * (MAS_Clusters-1))

} % end > 1 MAS cluster

Else % only one MAS cluster

{

Cluster_one_Keycode_Sessions = Total_MAS_Sessions_Licenses
ClusterN_keycode_Sessions = 0
Numb_Cluster_one_keycodes = 1
Numb_ClusterN_keycodes = 0

}

The customer now needs Numb_Cluster_one_keycodes of size Cluster_one_keycode_Sessions. If the number of MAS clusters (MAS_Clusters) is greater than one, an additional keycode of size ClusterN_keycode_Sessions will be required.

To verify the number of keycodes that must be generated:

Numb_Cluster_one_keycodes + Numb_ClusterN_keycodes = MAS_Clusters

To verify the size of the keycodes:

(Numb_Cluster_one_keycodes * Cluster_one_Keycode_Sessions) +
(Numb_ClusterN_keycodes * ClusterN_keycode_Sessions)
= Total_MAS_Sessions_Licenses

Chapter 16: Assigning loops and card slots in the Communication Server 1000E

Contents

This chapter contains the following topics:

Introduction

Calculating the number and assignment of cards and, relatedly, Media Gateways is an iterative procedure, because of specific capacity and usage requirements.

In an Avaya Communication Server 1000E (Avaya 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, and the Media Gateway Controller (MGC). These resources are available only to the Media Gateway (with optional Expander) that the circuit cards reside. Other services, such as Conference, are available as system resources but require Media Gateway-specific DSP resources in order to access them.

System capacities on page 207 and Resource calculations on page 249 describe the theoretical, traffic-based calculations used by Enterprise Configurator to estimate the required number of Media Cards and Media Gateways. This chapter describes the steps to allocate the cards to specific Media Gateways. The process can result in an increase in the required number of Media Cards and Media Gateways.

Note on terminology

The term Media Gateway refers to the Media Gateway 1010 (MG 1010) and Avaya CS 1000 Media Gateway 1000E (Avaya MG 1000E). The MG 1010 provides ten IPE slots. The Avaya MG 1000E provides four IPE slots.

Each MG 1000E can be connected to an optional Media Gateway Expander in order to increase capacity to eight IPE slots. In this chapter, the term MG 1000E includes the optional Media Gateway Expander, if equipped.

Loops and superloops

A fully expanded Avaya 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 Media Gateways. In such a case, the first Media Gateway is referred to as shelf 0 and the second as shelf 1.

A maximum of 1024 TNs (= 2 × 16 × 32) from two Media Gateways can be associated with a superloop.

Virtual superloops

There are no physical timeslots on Media Gateways. Timeslots are defined within virtual superloops that benefit from the nonblocking 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 Media Gateways. This expands the superloop's 120 timeslots to 1920 timeslots (= 16 × 120) to service a maximum of 1024 TNs in the address space. Media Gateways are therefore nonblocking 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 nonblocking superloops.

MGTDS and MGCONF loops

Media Gateway Tone and Conference configuration is configured in LD 17.

With MGTDS, you can configure two Media Gateway TDS loops. MGC-based Media Gateway lets 30 parties on each loop.

MGC-based Media Gateway capacity is 2 MGCONF loops with 30 parties on each loop.

PRI/PRI2/DTI/DTI2 loops

An MGC is required in slot 0 of any Media Gateway that will contain a PRI/PRI2/DTI/DTI2 card.

Each T1/E1 span consumes a loop, as well as a card slot.

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 90: Card slots in the MG 1000E chassis on page 381 summarizes the physical and logical card slots available in the MG 1000E chassis.

Table 91: Card slots in the MG 1010 chassis on page 382 summarizes the physical and logical card slots available in the MG 1010 chassis.

Table 92: Card slots in the MG 1000E cabinet on page 383 summarizes the physical and logical card slots available in the MG 1000E cabinet.

Table 90: Card slots in the MG 1000E chassis

Slot number		Used for	Comment
Physical	Logical		
MG 1000E chassis			
0	0	MGC	Dedicated card slot
1–4	1–4	• Media Cards • Digital line cards • Digital trunk cards • Analog line cards • Analog trunk cards • Application cards • CP PM cards	
n/a	5–6	n/a	Not supported

Slot number		Used for	Comment
Physical	Logical		
Media Gateway Expander			
7–10	7–10	• Media Cards • Digital line cards • Analog line cards • Analog trunk cards • Application cards • CP PM 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		
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 91: Card slots in the MG 1010 chassis

Slot number		Used for	Comment
Physical	Logical		
MG 1010 chassis			
0	0	MGC (NTDW98)	Dedicated card slot
1–10	1–10	• Media Cards • Digital line cards • Digital trunk cards • Analog line cards	You can use NTDW61 CP PM cards in IPE slots 1-10. CP PM cards in slots 1-10 require the NTAK19 cable kit for serial connections.

Slot number		Used for	Comment
Physical	Logical		
		• Analog trunk cards • Application cards • CP PM cards	
n/a	21	MGU	Dedicated card slot
n/a	22	CP PM cards (NTDW99)	Dedicated card slot
n/a	23	CP PM cards (NTDW99)	Dedicated card slot
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		
n/a	14	DTRs	Required if any analog terminals or trunks are equipped in the MG 1010.
n/a	15	DTRs	
n/a	15	MF tone detectors	Must be provided on each MG 1010 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 92: Card slots in the MG 1000E cabinet

Slot number		Used for	Comment
Physical	Logical		
MG 1000E cabinet			
0	0	MGC	Dedicated card slot
1–9	1–9	• Media Cards • Digital line cards • Digital trunk cards • Analog line cards	

Slot number		Used for	Comment
Physical	Logical		
		• 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.
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

Media Gateways are nonblocking with respect to timeslots (see Virtual superloops on page 380). Blocking can occur only if a Media Gateway 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 card in slot 0 of each Media Gateway.
2. There must be at least 32 DSP ports in each Media Gateway.

Important:

DSP resources cannot be shared between Media Gateways. Therefore, each Media Gateway must contain sufficient DSP resources required by the equipment configured in that Media Gateway.

3. There must be at least one TDS loop in each Media Gateway.
4. Allocate the users and Media Cards for dedicated DSPs first, then fill remaining empty slots in Media Gateways with other IPE cards.

Important:

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 Media Gateways.

Provision resources in the following order:

a. Conference on page 385
b. TDS on page 386
c. Broadcast circuits on page 387
d. Other service circuits on page 389
e. TDM telephones and TDM agents on page 389
f. Consoles on page 390
g. Standard telephones on page 391

Conference

Each Media Gateway provides up to 60 conference circuits (ports), which can be used to form conferences of up to 30 parties each. The MGC card has 60 conference circuits (2 loops). Users can configure 2 conference loops on each MGC-based Media Gateway, with each loop providing 30 conference circuits.

The conference circuits are available to all Media Gateways 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 Media Gateway 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 on page 265).

For nonblocking access, provide an equal number of DSP ports and conference ports. In other words, provide one 32-port Media Card for every defined conference loop.

Media Gateway and DSP calculations for Conference

1. Calculate the number of Media Gateways required for Conference based on the number of conference circuits needed, in multiples of 60:

   Number of Media Gateways = ROUNDUP(Number of conference circuits ÷ 60)

2. Calculate the number of DSPs required for Conference based on the number of conference circuits needed, in multiples of 32:

   Provide 32 ports of DSP for every defined conference loop.

Examples

1. 30 conference circuits are needed:
   - One Media Gateway has Conference configured. All other Media Gateways do not have conference circuits provisioned.
   - The Media Gateway with Conference requires 32 DSP ports to support the service.
2. 33 conference circuits are needed:
   - One Media Gateway has Conference configured. All other Media Gateways do not have conference circuits provisioned.
   - The Media Gateway with Conference requires 64 DSP ports to support the service.
3. 100 conference circuits are needed:
   - Two Media Gateways have Conference configured. All other Media Gateways do not have conference circuits provisioned.
   - Each Media Gateway with Conference requires 64 DSP ports to support the service.

TDS

A minimum of one TDS loop is required in each Media Gateway. The TDS circuits are provided by the MGC card. If additional TDS circuits are required in any Media Gateway, a second TDS loop can be configured in it.

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 1-9 of a Media Gateway.

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 93: Example of timeslot sharing in a superloop on page 387, 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 an Avaya Integrated Recorded Announcer card in one Media Gateway 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 Media Gateway is not used. Table 93: Example of timeslot sharing in a superloop on page 387 illustrates the strategy.

Table 93: Example of timeslot sharing in a superloop

Media Gateway 0				Media Gateway 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.]

Media Gateway 0				Media Gateway 1			
l	**s**	**c**	**Card**	**l**	**s**	**c**	**Card**
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.]
0	0	9	Media Card for RAN/Music	0	1	9	[Available for use.]
0	0	10	[Leave empty — all DSPs in this Media Gateway 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 Media Gateway 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.

Media Gateway 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 Media Gateway to support all of the functions within that chassis or cabinet. If 128 ports of CallPilot are provided, 128 DSPs are required for CallPilot usage. If controlled broadcast is used with 120 connections, then only 8 DSPs would be left for access to all remaining CallPilot 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.

Media Gateway and DSP calculations for broadcast circuits

Since each Recorded Announcer card can broadcast to up to 120 listeners, each card requires a maximum of 120 DSP ports for all music or RAN source channels for that card. Each Media Gateway 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 Media Gateway.
- If 32 < (RAN CON + MUS CON) <= 120, allocate 4 Media Cards + 1 Recorded Announcer card to the same Media Gateway.

Other service circuits

The list of other service circuits is extensive. It includes, amongst others, cards such as Avaya Integrated applications, Avaya CallPilot, digital trunks, 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 on page 393.)

Consoles

Each Avaya 2250 Attendant Console and PC Console requires two TNs (originating and terminating) on an XDLC card and one Aux TN (for supervisor function). Avaya 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).

IP Attendant Consoles

Each IP Attendant 3260 in the system requires four SIP ports. You must ensure that there are enough SIP ports to support the intended number of consoles.

Traffic estimation

You only need to perform traffic estimation calculations if the overall bandwidth between the IP Attendant Consoles and the registered Media Services server is less than 20 Mbps. This applies to all deployments.

The traffic estimation calculations shown in the table below are only applicable to the traffic between the IP Attendant Console and the Media Services server.

Table 94: Traffic estimation calculations for IP Attendant Console

Attendants	Estimated Traffic Load (Mbps)	Load Requirements
1	0.07	0.1
2	0.14	0.15
3	0.21	0.25
4	0.28	0.3
5	0.35	0.4
10	0.7	0.75

Attendants	Estimated Traffic Load (Mbps)	Load Requirements
15	1.05	1.1
20	1.4	1.4
30	2.1	2.1
40	2.8	2.8
50	3.5	3.5
60	4.2	4.2
63	4.41	4.4

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 Media Gateways 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 Media Gateways 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.
   - There are a total of 10 card slots available in each MG 1010.
   - If 1 card slot is used by a Media Card, a maximum of 9 line cards, or 144 telephones (9 × 16 ports), can be added to the MG 1010.
   - A Media Card with 32 DSPs supports 794 CCS. This is the traffic capacity of this particular Media Gateway.
   - A capacity limit of 794 CCS means each MG 1000E based telephone must generate less than 7 CCS, on average (794 ÷ 112).
   - A capacity limit of 794 CCS means each MG 1010 based telephone must generate less than 5.5 CCS, on average (794 ÷ 144).

   For average traffic of more than 7 CCS per telephone, use Table 95: Maximum number of Media Cards, line cards, and telephones in a Media Gateway on

page 392 to determine the number of Media Cards and telephones that can be assigned to an MG 1000E.

Table 95: Maximum number of Media Cards, line cards, and telephones in a Media Gateway

CCS per telephone	Media Cards	Line cards	Telephones*
<=5.5	1	9	144
<= 7.0	1	7	112
<= 8.0	1	6	96
<= 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 96: Maximum number of Media Cards, trunk cards, and trunks in a Media Gateway on page 392.

Table 96: Maximum number of Media Cards, trunk cards, and trunks in a Media Gateway

CCS per telephone	Media Cards	Trunk cards	Trunks*
<= 11.0	1	9	72
<= 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 a Media Gateway, calculate the total CCS for the number of lines and trunks. Then use Table 97: Traffic capacity of Media Cards (Erlang B at P.01) on page 392 to identify the number of Media Cards required to support that CCS rate.

Table 97: 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 Media Gateway. 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 98: Number of Media Cards, line cards, and ACD agents per superloop on page 393 describes two alternative recommended configurations to provide nonblocking ACD access to DSP ports. Typically, 33 CCS per agent telephone is engineered. However, in a nonblocking configuration, up to 36 CCS per agent is allowed.

Table 98: Number of Media Cards, line cards, and ACD agents per superloop

Total number of agents*	Media Gateway (shelf 0)			Media Gateway (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 Media Gateways developed in accordance with this chapter.
2. Go back to the Enterprise Configurator theoretical calculations and increment the order requirements to match the modified configuration.
3. Produce the final Enterprise Configurator configuration report. Enterprise Configurator output includes the following:
    - card locations
    - DSP allocations (where Media Card utilization is less than 100%)
    - Media Gateways that must have conference circuits provisioned

- Media Gateways that must not have conference circuits provisioned

Chapter 17: Provisioning

Contents

This chapter contains the following topics:

Introduction

This section provides a high-level overview of the steps required to determine general equipment requirements. Consult your Avaya representative and use a configuration tool, such as Enterprise Configurator, to fully engineer a system.

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 99: Example of manual calculation of CCS/T on page 398).

Use an appropriate trunking table (see Reference tables on page 413) 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 99: Example of manual calculation of CCS/T on page 398 provides an example of the manual calculation.

Table 99: 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 on page 415). 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 on page 415 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

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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 Resource calculation worksheets on page 403 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 card and optional, additional XDTR cards, see DTR on page 221 .

In the Avaya Communication Server 1000E (Avaya CS 1000E), DTRs are not system-wide resources. They support only the telephones and trunks in the Media Gateway that they reside in. See DTR on page 221 for the calculations to determine overall DTR traffic and refer to reference tables Digitone receiver requirements Model 1 on page 418 through Digitone receiver requirements Model 4 on page 421 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 on page 418 through Digitone receiver requirements Model 4 on page 421 are based on some common PBX traffic measurements.

Model 1

Digitone receiver requirements Model 1 on page 418 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 on page 419 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 on page 420 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 on page 421 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

Resource calculation worksheets on page 403 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. For additional rules and tips to distribute the Media Cards amongst the Media Gateways in order to determine final Media Card requirements, see Assigning loops and card slots in the Communication Server 1000E on page 379 .

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 on page 271 .

Step 10: Provision conference/TDS loops

The MGC card provides conference/TDS functions. For information about provisioning these functions within each Media Gateway, see Conference on page 385 and TDS on page 386 .

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 Media Gateway. See Assigning loops and card slots in the Communication Server 1000E on page 379.

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.

Important:

Another step you want to consider at this point is system security. For more information, see *Avaya Access Control Management Reference, NN43001-602*.

Resource calculation worksheets

The following lists information you require to determine card placement, DSP usage, and Signaling Server requirements.

Input parameters	Input configuration data
$Telephone_{CCS}$ – CCS for each standard telephone	Number of TDM telephones (analog and digital), both blocking and nonblocking
TRK_{CCS} – CCS for each trunk	Number of UNIStim IP Phones
$NBtelephone_{CCS}$ – CCS for each nonblocking telephone	Number of SIP IP Phones
ACD_{CCS} – CCS for each ACD agent	Number of DECT telephones, including SIP-DECT
R_I – intraoffice calls ratio	Number of TDM ACD agents
R_T – tandem calls ratio	Number of UNIStim IP ACD agents
I – incoming calls to total calls ratio	Number of TDM trunks
O – outgoing calls to total calls ratio	Number of SIP Virtual Trunks (estimated)
r_{CON} – Conference loop to traffic loop ratio	Number of H.323 Virtual Trunks (estimated)
Hold time in seconds (AHT_{XX})for telephone to telephone, trunk to trunk, telephone to trunk, trunk to telephone	

Use the calculations in Table 100: Worksheet A: Resource calculation procedure on page 403 for input into the Table 102: Worksheet C: Virtual Trunk calculation on page 407, and for input into the Real time calculation worksheets on page 408. Worksheet A is not required for input into the DSP calculations for a Communication Server 1000E system.

Table 100: Worksheet A: Resource calculation procedure

Item	Calculation formula
(1) TDM telephone CCS (L_{TDM})	= ((number of analog telephones + number of digital telephones + number of line-side T1/E1 ports) × _____ $Telephone_{CCS}$) + (number of nonblocking telephones × _____ $NBtelephone_{CCS}$)
(2) UNIStim IP telephone CCS (L_{IP})	= (number of UNIStim IP telephones - number of IP ACD agents) × ______ $Telephone_{CCS}$
(3) TDM ACD agent CCS (L_{ACD})	= (number of TDM ACD agents) × ______ ACD_{CCS}

Item	Calculation formula
(4) UNIStim IP ACD agent CCS (L_{ACDIP})	= (number of UNIStim IP ACD agents) × ______ ACD_{CCS}
(5) DECT telephone CCS (L_{DECT})	= (number of DECT telephones + number of SIP-DECT telephones) × ______ $Telephone_{CCS}$
(6) IP 802.11 Wireless telephone CCS (L_{IPW})	= (number of 802.11 Wireless telephones) × ______ $Telephone_{CCS}$
(7) SIP Line IP telephone CCS (L_{SIPL})	= (number of SIP Line IP telephones) × ______ $Telephone_{CCS}$
(8) ACD CCS adjustment for TDM agents (ACD_{adj})	= number of TDM ACD agents × ______ $NBtelephone_{CCS}$
(9) Total line CCS (L_{CCS})	= $L_{TDM} + L_{IP} + L_{ACD} + L_{ACDIP} + L_{DECT} + L_{IPW} + L_{SIPL} - ACD_{adj}$ = (1) + (2) + (3) + (4) + (5) + (6) + (7) - (8)
(10) TDM trunk CCS (T_{TDM})	= Number of TDM trunks × ____ TRK_{CCS}
(11) SIP Virtual Trunk CCS (SVT_{CCS})	= Number of SIP Virtual Trunks × ____ TRK_{CCS}
(12) H.323 Virtual Trunk CCS (HVT_{CCS})	= Number of H.323 Virtual Trunks × ____ TRK_{CCS}
(13) Total Virtual Trunk CCS (VT_{CCS})	= $SVT_{CCS} + HVT_{CCS}$ = (11) + (12)
(14) Total trunk CCS (T_{TCCS})	= $T_{TDM} + VT_{CCS}$ = (10) + (13)
(15) Total system CCS (T_{CCS})	= $L_{CCS} + T_{TCCS}$ = (9) + (14)
(16) Percentage H.323 trunk CCS of total Virtual Trunk CCS (V_H)	= $HVT_{CCS} \div VT_{CCS}$ = (12) ÷ (13)
(17) Percentage SIP trunk CSS of total Virtual Trunk CCS (V_S)	= $SVT_{CCS} \div VT_{CCS}$ = (11) ÷ (13)
(18) Percentage Virtual Trunk CSS of total trunk CCS (V)	= $VT_{CCS} \div T_{TCCS}$ = (13) ÷ (14)
(19) UNIStim IP CSS to total telephone CCS ratio (P_U)	= $(L_{IP} + L_{ACDIP}) \div L_{CCS}$ = ((2) + (4)) ÷ (9)
(20) SIP Line IP CSS to total telephone CCS ratio (P_S)	= $L_{SIPL} \div L_{CCS}$ = (7) ÷ (9)
(21) IP CSS to total telephone CCS ratio (P_{IP})	= $P_U + P_S$ = (19) + (20)
(22) Weighted average holding time (WAHT)	= $(R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$

Item	Calculation formula
(23) Total calls (T_{CALL})	$= 0.5 \times T_{CCS} \times 100 \div WAHT$ $= 0.5 \times (15) \times 100 \div (22)$
(24) Intraoffice calls (C_{SS})	$= T_{CALL} \times R_I$
• (a) Intraoffice UNIStim IP to UNIStim IP calls (C_{2IP})	$= C_{SS} \times P_U \times P_U$
• (b) Intraoffice UNIStim IP to TDM telephone calls (C_{1IP})	$= C_{SS} \times 2 \times P_U \times (1 - P_{IP})$
• (c) Intraoffice TDM telephone to TDM telephone calls (C_{NoIP})	$= C_{SS} \times (1 – P_{IP})^2$
• (d) Intraoffice SIP Line to SIP Line calls (C_{2sip})	$= C_{SS} \times P_S \times P_S$
• (e) Intraoffice SIP Line to UNIStim IP calls ($C_{2sipuip}$)	$= C_{SS} \times P_S \times P_U$
• (f) Intraoffice SIP Line to TDM calls (C_{1sip})	$= C_{SS} \times 2 \times P_S \times (1 - P_{IP})$
(25) Tandem calls (C_{TT})	$= T_{CALL} \times R_T$
• (a) Tandem VT to TDM trunk calls (C_{T1VT})	$= C_{TT} \times 2 \times V \times (1 - V)$
• (b) Tandem TDM trunk to TDM trunk calls (C_{T2NoVT})	$= C_{TT} \times (1 - V)^2$
• (c) Tandem VT (H323) to VT (SIP) calls (C_{T2HS})	$= C_{TT} \times V^2 \times V_H \times V_S \times 2 \times 2$
(26) Originating / Outgoing calls (C_{ST})	$= T_{CALL} \times O$ (outgoing ratio)
• (a) UNIStim IP to VT calls (C_{STIV})	$= C_{ST} \times P_U \times V$
• (b) UNIStim IP to TDM calls (C_{STID})	$= C_{ST} \times P_U \times (1 - V)$
• (c) TDM telephone to VT calls (C_{STDV})	$= C_{ST} \times (1 - P_{IP}) \times V$
• (d) TDM telephone to TDM trunk calls (C_{STDD})	$= C_{ST} \times (1 - P_{IP}) \times (1 - V)$
• (e) SIP Line to VT calls (C_{STSV})	$= C_{ST} \times P_S \times V$

Item	Calculation formula
• (f) SIP Line to TDM trunk calls (C_{STSD})	= $C_{ST} \times P_S \times (1 - V)$
(27) Terminating / Incoming calls (C_{TS})	= $C_{TS} \times I$ (incoming ratio)
• (a) VT to TDM telephone calls (C_{TSVD})	= $C_{TS} \times V \times (1 - P_{IP})$
• (b) VT to UNIStim IP telephone calls (C_{TSVI})	= $C_{TS} \times V \times P_U$
• (c) TDM trunk to UNIStim IP telephone calls (C_{TSDI})	= $C_{TS} \times (1 - V) \times P_U$
• (d) TDM trunk to TDM telephone calls (C_{TSDD})	= $C_{TS} \times (1 - V) \times (1 - P_{IP})$
• (e) VT to SIP Line telephone calls (C_{TSVS})	= $C_{TS} \times V \times P_S$
• (f) TDM trunk to SIP Line telephone calls (C_{TSDS})	= $C_{TS} \times (1 - V) \times P_S$

Table 101: Worksheet B: Detailed DSP and Media Card calculation for Media Gateway

Item	Calculation formula
Calculate for each Media Gateway	Media Gateway location ____________________________
(1) DSP ports for nonblocking ports (telephones, ACD agents, consoles, digital trunks, and analog trunks)	= Number of trunk ports + number of telephones + [4 × number of standard consoles] + [12 × number of nonblocking consoles]
(2) Media Gateway CCS for general traffic (CCS_{DSP})	= Number of standard ports × $telephone_{CCS}$
(3) DSP ports for general traffic	= lookup the CCS_{DSP} value (2) in Table 57: Erlang B and Poisson values, in 32-port increments on page 264 for the number of DSP ports
(4) DSP ports for conference	= $Total\_telephones \times P_{IP} \times r_{CON} \times 0.4$ = Number of configured conference loops × 32 (maximum of 2 conference loops for each Media Gateway)
(5) DSP ports for applications	= (a) + (b) + (c) + (d) + (e) + (f) + (g) + (h)

Item	Calculation formula
• (a) CallPilot	= Number of CallPilot ports
• (b) MIRAN	= Number of MIRAN listeners (maximum of 120 listeners for each Media Gateway)
• (c) Integrated Conference Bridge	= Number of ICB ports
• (d) Integrated Conference Director	= Number of IPCB ports
• (e) Integrated Call Assistant	= Number of ICA ports
• (f) Hospitality Integrated Voice Service	= Number of IVS ports
• (g) BRI	= Number of BRI ports × 2
• (h) Agent greeting ports	= Number of Agent greeting ports
(6) Total DSP ports	= (1) + (3) + (4) + (5)
(7) Extra Media Cards required *	= Roundup [[(6) - 128] ÷ 32]
* Number for extra Media Cards assumes each Media Gateway contains 128 DSP ports on the MGC card. If extra Media Cards are required, then you may need to move other cards out of the Media Gateway to provide enough available slots to hold more Media Cards.	

Table 102: Worksheet C: Virtual Trunk calculation

Call type	Calculation formula
(1) Virtual Trunk calls (C_{VT})	= $C_{T1VT} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} + C_{T2HS} + C_{STSV} + C_{TSVS}$
(2) SIP Virtual Trunk calls	$C_{VT} \times V_S$
(3) H.323 Virtual Trunk calls	$C_{VT} \times V_H$
(4) Virtual Trunk CCS (CCS_{VT})	C_{VT} × WAHT ÷ 100
(5) Number of Virtual Trunks	Roundup (CCS_{VT} ÷ 5084 × 192)
(6) Virtual Trunk traffic in erlangs	Roundup (CCS_{VT} ÷ 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%), Avaya 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.	

Real time calculation worksheets

The variable values in many of the following tables are calculated in Table 100: Worksheet A: Resource calculation procedure on page 403.

Table 103: Worksheet D-1: Basic call EBC calculation

Item	Calculation formula
(1) Intraoffice UNIStim IP to UNIStim IP call penetration factor	P_UIPtoUIP = $C_{2IP} \div T_{CALL}$
(2) Intraoffice UNIStim IP to TDM telephone calls penetration factor	P_UIPtoL = $C_{1IP} \div T_{CALL}$
(3) Intraoffice TDM telephone to TDM telephone calls penetration factor	P_LtoL = $C_{NoIP} \div T_{CALL}$
(4) Intraoffice SIP Line to SIP Line calls penetration factor	P_SIPtoSIP = $C_{2sip} \div T_{CALL}$
(5) Intraoffice SIP Line to UNIStim IP calls penetration factor	P_SIPtoUIP = $C_{2sipuip} \div T_{CALL}$
(6) Intraoffice SIP Line to TDM telephone calls penetration factor	P_SIPtoL = $C_{1sip} \div T_{CALL}$
(7) Tandem Virtual Trunk to TDM trunk calls penetration factor	P_VTtoTr = $C_{T1VT} \div T_{CALL}$
(8) Tandem TDM trunk to TDM trunk calls penetration factor	P_TrtoTr = $C_{T2NoVT} \div T_{CALL}$
(9) Tandem VT (H323) to VT (SIP) calls penetration factor	P_VhtoVs = $C_{T2HS} \div T_{CALL}$
(10) UNIStim IP to VT calls penetration factor	P_UIPtoVT = $C_{STIV} \div T_{CALL}$
(11) UNIStim IP to TDM trunk calls penetration factor	P_UIPtoTr = $C_{STID} \div T_{CALL}$
(12) TDM telephone to VT calls penetration factor	P_LtoVT = $C_{STDV} \div T_{CALL}$
(13) TDM telephone to TDM trunk calls penetration factor	P_LtoTr = $C_{STDD} \div T_{CALL}$
(14) SIP Line to VT calls penetration factor	P_SIPtoVT = $C_{STSV} \div T_{CALL}$
(15) SIP Line to TDM trunk calls penetration factor	P_SIPtoTr = $C_{STSD} \div T_{CALL}$
(16) VT to TDM telephone calls penetration factor	P_VTtoL = $C_{TSVD} \div T_{CALL}$

Item	Calculation formula
(17) VT to UNIStim IP telephone calls penetration factor	$P\_VTtoUIP = C_{TSVI} \div T_{CALL}$
(18) TDM trunk to UNIStim IP telephone penetration factor	$P\_TrtoUIP = C_{TSDI} \div T_{CALL}$
(19) TDM trunk to TDM telephone penetration factor	$P\_TrtoL = C_{TSDD} \div T_{CALL}$
(20) VT to SIP Line telephone calls penetration factor	$P\_VTtoSIP = C_{TSVS} \div T_{CALL}$
(21) TDM trunk to SIP Line telephone penetration factor	$P\_TrtoSIP = C_{TSDS} \div T_{CALL}$
(22) Weighted average penetration factor	$PF = (P\_UIPtoUIP \times f_1) + (P\_UIPtoL \times f_2) + (P\_LtoL \times f_3) + (P\_VTtoTr \times f_4) + (P\_TrtoTr \times f_5) + (P\_VhtoVs \times f_6) + (P\_UIPtoVT \times f_7) + (P\_UIPtoTr \times f_8) + (P\_LtoVT \times f_9) + (P\_LtoTr \times f_{10}) + (P\_VTtoL \times f_{11}) + (P\_VTtoUIP \times f_{12}) + (P\_TrtoUIP \times f_{13}) + (P\_TrtoL \times f_{14}) + (P\_SIPtoSIP \times f_{15}) + (P\_SIPtoUIP \times f_{16}) + (P\_SIPtoL \times f_{17}) + (P\_SIPtoVT \times f_{18}) + (P\_SIPtoTr \times f_{19}) + (P\_VTtoSIP \times f_{20}) + (P\_TrtoSIP \times f_{21})$
(23) Error_term (basic features: forward/ transfer/conference/waiting)	Error_term = 0.25
(24) System EBC	$SEBC = (T_{CALL} \times (1 + PF + Error\_term))$

Table 104: Worksheet D-2: Feature and application EBC calculation

Item	Calculation formula
ACD calls	$C_{ACD} = (L_{ACD} + L_{IPACD}) \times 100 \div AHT_{AGT}$
ACD	$ACDEBC = C_{ACD} \times (1 - \%Symposium) \times f_{ACD}$
Symposium	$SymposiumEBC = \%Symposium \times C_{ACD} \times f_{SYM}$
CallPilot	$CallPilotEBC = (CP1 + CP2) \times 100 \div AHT_{CP} \times f_{CP}$
Internal CDR	$InternalCDR\_EBC = C_{SS} \times f_{ICDR}$
Incoming CDR	$IncomingCDR\_EBC = C_{TS} \times f_{CCDR}$
Outgoing CDR	$OutgoingCDR\_EBC = C_{ST} \times f_{OCDR}$
Tandem CDR	$TandemCDR\_EBC = C_{TT} \times f_{TAN}$
Integrated Conference Bridge	MICB_EBC = number of Integrated Conference Bridge ports × $Appl_{CCS} \times 100 \div AHT_{MICB} \times f_{MICB}$

Item	Calculation formula
Integrated Recorded Announcer	MIRAN_EBC = number of Integrated Recorded Announcer ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIRAN} × f_{MIRAN}
Integrated Call Director	MIPCD_EBC = number of Integrated Call Director ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIPCD} × f_{MIPCD}
Integrated Call Announcer	MICA_EBC = number of Integrated Call Announcer ports × $Appl_{CCS}$ × 100 ÷ AHT_{MICA} × f_{MICA}
Hospitality Integrated Voice Services	MIVS_EBC = number of Hospitality Integrated Voice Services ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIVS} × f_{MIVS}
BRI	BRI_EBC = number of BRI users × $Telephone_{CCS}$ × 100 ÷ AHT_{BRI} × f_{BRI}
MDECT	MDECT_EBC = LDECT × 100 ÷ WAHT × f_{DECT}
CPND	CPND_EBC = $(C_{SS} + C_{TS}) \times f_{CPND}$
Converged Desktop	CD_EBC = $(C_{SS} \times 0.1 + C_{TT} + C_{ST} + C_{TS}) \times r_{DTP} \times f_{DTP}$
Microsoft Converged Office	MO_EBC = $(C_{SS} \times 0.1 + C_{TT} + C_{ST} + C_{TS})$ × mop × f_{MO}
IP Security	IPSEC_EBC = $(C_{SS} + C_{TT} + C_{ST} + C_{TS}) \times P_{IP}$ × IPSEC_P × f_{ipsec}
MC3100	MC3100_EBC = $(C_{SS} + C_{TT} + C_{ST} + C_{TS})$ × MC3100_P × f_{mc3100}
MobileX	MobileX_EBC = $(C_{SS} + C_{TT} + C_{ST} + C_{TS})$ × MobileX_P × $f_{mobileX}$
ELC	ELC_EBC = (C_{SS} × ELC_P × f_{ELC}
Features and Applications EBC	FAEBC = ACD_EBC + Symposium_EBC + CallPilot_EBC + InternalCDR_EBC + IncomingCDR_EBC + OutgoingCDR_EBC + TandemCDR_EBC + MICB_EBC + MIRAN_EBC + MIPCD_EBC + MICA_EBC + MIVS_EBC + BRI_EBC + MDECT_EBC + CPND_EBC + CD_EBC + MO_EBC + IPSEC_EBC + MC3100_EBC + MobileX_EBC + ELC_EBC

Table 105: Worksheet D-3: Real Time Usage calculation

Item	Calculation formula
Real Time Usage	RTU = (SEBC + FAEBC) ÷ Rated_EBC × 100

Item	Calculation formula
	where Rated_EBC is the rated capacity of the CPU from Table 43: Real time capacity (EBC) by system on page 227

Appendix A: Reference tables

List of tables

Trunk traffic Erlang B with P.01 Grade-of-Service

Table 106: 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

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
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
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

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
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 107: Trunk traffic Poisson 1 percent blocking

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	41	993	81	2215	121	3488	161	4786
2	5.4	42	1023	82	2247	122	3520	162	4819
3	15.7	43	1052	83	2278	123	3552	163	4851
4	29.6	44	1082	84	2310	124	3594	164	4884
5	46.1	45	1112	85	2341	125	3616	165	4917
6	64	46	1142	86	2373	126	3648	166	4549
7	84	47	1171	87	2404	127	3681	167	4982
8	105	48	1201	88	2436	128	3713	168	5015
9	126	49	1231	89	2467	129	3746	169	5048
10	149	50	1261	90	2499	130	3778	170	5081
11	172	51	1291	91	2530	131	3810	171	5114
12	195	52	1322	92	2563	132	3843	172	5146
13	220	53	1352	93	2594	133	3875	173	5179
14	244	54	1382	94	2625	134	3907	174	5212
15	269	55	1412	95	2657	135	3939	175	5245
16	294	56	1443	96	2689	136	3972	176	5277
17	320	57	1473	97	2721	137	4004	177	5310
18	346	58	1504	98	2752	138	4037	178	5343
19	373	59	1534	99	2784	139	4070	179	5376
20	399	60	1565	100	2816	140	4102	180	5409

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
21	426	61	1595	101	2847	141	4134	181	5442
22	453	62	1626	102	2879	142	4167	182	5475
23	480	63	1657	103	2910	143	4199	183	5508
24	507	64	1687	104	2942	144	4231	184	5541
25	535	65	1718	105	2974	145	4264	185	5574
26	562	66	1749	106	3006	146	4297	186	5606
27	590	67	1780	107	3038	147	4329	187	5639
28	618	68	1811	108	3070	148	4362	188	5672
29	647	69	1842	109	3102	149	4395	189	5705
30	675	70	1873	110	3135	150	4427	190	5738
31	703	71	1904	111	3166	151	4460	191	5771
32	732	72	1935	112	3198	152	4492	192	5804
33	760	73	1966	113	3230	153	4525	193	5837
34	789	74	1997	114	3262	154	4557	194	5871
35	818	75	2028	115	3294	155	4590	195	5904
36	847	76	2059	116	3326	156	4622	196	5937
37	876	77	2091	117	3359	157	4655	197	5969
38	905	78	2122	118	3391	158	4686	198	6002
39	935	79	2153	119	3424	159	4721	199	6035
40	964	80	2184	120	3456	160	4754	200	6068
For trunk traffic greater than 6068 CCS, allow 30.34 CCS per trunk.									

Trunk traffic Poisson 2 percent blocking

Table 108: 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

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
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
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 109: 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 110: 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 111: Digitone receiver requirements Model 3

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
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 112: 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
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 113: Digitone receiver load capacity 6 to 15 second holding time

Number 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
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

Number of DTRs	Average holding time in seconds									
	6	7	8	9	10	11	12	13	14	15
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 114: 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

Number of DTRs	Average holding time in seconds									
	16	17	18	19	20	21	22	23	24	25
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
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

Number of DTRs	Average holding time in seconds									
	16	17	18	19	20	21	22	23	24	25
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 115: 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 116: 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 117: 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

DTR CCS	DTR ports	DTR CCS	DTR ports
1330–1365	50	2101–2135	72
1366–1400	51	2136–2170	73
1401–1435	52	2171–2205	74
1436–1470	53	2206–2240	75
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.			

Appendix B: Blank ELAN Bandwidth tables

Table 118: MG1000 Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
10				
9				
8				
7				
6				
5				
4				
3				
2				
1				
0				
		Totals		
		Chassis Total		
		With IPSec ON (1.3 * chassis total)		

Table 119: MG1010 Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
10				
9				
8				
7				
6				
5				

4				
3				
2				
1				
0				
23				
22				
Totals				
Chassis Total				
With IPSec ON (1.3 * chassis total)				

Table 120: MG PRI Gateway Media Gateway ELAN Estimation

Slot	Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
2				
1				
0				
Totals				
Chassis Total				
With IPSec ON (1.3 * chassis total)				

Table 121: MGXPEC ELAN Estimation

Slot		Card Type	Estimated load (Normal or High)	Idle Traffic (Bits/sec)	Traffic Load (Bits/sec)
15					
14					
13					
12					
11					
10					
9					
8					

		MGXPEC		12000	0
7					
6					
5					
4					
3					
2					
1					
0					
			Chassis Total		
			With IPSec ON (1.3 * chassis total)		