Communication Server 1000M and Meridian 1 Large System Planning and Engineering
Avaya Communication Server 1000

7.5
NN43021-220, 05.06
April 2012

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Chapter 1: New in this release

This chapter outlines the new or updated features in this document for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.5. This chapter also provides information about this document.

Contents

This chapter contains the following topics:

Features on page 15

Other on page 15

Features

The Extend Local Calls (ELC) feature for CS 1000M requires DSP resources. DSP ports are required to support TDM to TDM calls using the ELC feature.

The QPC414 network card content is removed from this document for Release 7.5. The QPC414 network cards are manufacturer discontinued.

Other

Revision history

April 2012	Standard 05.06. This document is up-issued to include information about the surge-suppression cable for certain trunk cards.
March 2012	Standard 05.05. This document is up-issued to include updates to CSQI/CSQO limits.

July 2011	Standard 05.04. This document is up-issued to include an update to the Signaling Server capacities section. MAS cph details have been added.
June 2011	Standard 05.03. This document is updated to include the Avaya Common Server (HP DL360 G7).
February 2011	Standard 05.02. This document is up-issued to remove legacy feature and hardware content that is no longer applicable to or supported by Communication Server 1000 systems.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 05.01. This document is issued to support Avaya Communication Server 1000 Release 7.5.
February 2012	Standard 04.03. This document is up-issued to include updates to CSQI/CSQO limits.
August 2010	Standard 04.02. This document is up-issued to include planning and engineering capacity updates, and Signaling Server algorithm updates 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.
October 2009	Standard 03.05. This document is up-issued to support the Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card.
July 2009	Standard 03.04. This document is up-issued to update the temperature and humidity control section in the Preparing a system installation plan 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 update the System memory section in the Common Equipment (Core) and Table 47 in Memory Size chapter.
September 2008	Standard 02.06. This document is up-issued to reflect changes in technical content to the maximum limits for Media Card and DSPs.
September 2008	Standard 02.05. This document is up-issued to update System Capacities chapter.
August 2008	Standard 02.04. This document is up-issued to update System Capacities chapter.
May 2008	Standard 02.03. This document is up-issued to support technical corrections.

February 2008	Standard 02.02. This document is up-issued with corrections to real-time factors, software release degradation factors, and COTS servers running NRS on Linux cannot support signaling server applications.
December 2007	Standard 02.01. This document is issued to support Nortel Communication Server 1000 Release 5.5.
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 1000M and Meridian 1 Large System Planning and Engineering , NN43021-220*and contains information previously contained in the following legacy document, now retired: *Communication Server 1000M and Meridian 1: Large System Planning and Engineering, 553-3021-120*.

Chapter 2: Customer service

Visit the Avaya Web site to access the complete range of services and support that Avaya provides. Go to www.avaya.com or go to one of the pages listed in the following sections.

Navigation

- Getting technical documentation on page 19
- Getting product training on page 19
- Getting help from a distributor or reseller on page 19
- Getting technical support from the Avaya Web site on page 20

Getting technical documentation

To download and print selected technical publications and release notes directly from the Internet, go to www.avaya.com/support.

Getting product training

Ongoing product training is available. For more information or to register, go to www.avaya.com/support. From this Web site, locate the Training link on the left-hand navigation pane.

Getting help from a distributor or reseller

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

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

Contents

This chapter contains the following topics:

Introduction on page 21

Engineering a new system on page 22

Engineering a system upgrade on page 22

Communication Server 1000 task flow on page 24

Other resources on page 26

Introduction

Warning:

Before a Large 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 in which all components work within their capacity limits during the busy hour.

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

Engineering a new system

Figure 1: Engineering a new system on page 23 illustrates a typical process for installing a new system. The agent expected to perform each step of the process is listed to the right of the block. The highlighted block is the subject of this document. It is further illustrated in Figure 2: Engineering a new system on page 24.

Engineering a system upgrade

In cases of major upgrades or if current resource usage levels are not known, it is recommended that the complete engineering process be followed, as described in the previous section.

If minor changes are being made, the incremental capacity impacts can be calculated and added to the current resource usage levels. The resulting values can then be compared to the capacity chart to determine whether the corresponding capacity has been exceeded.

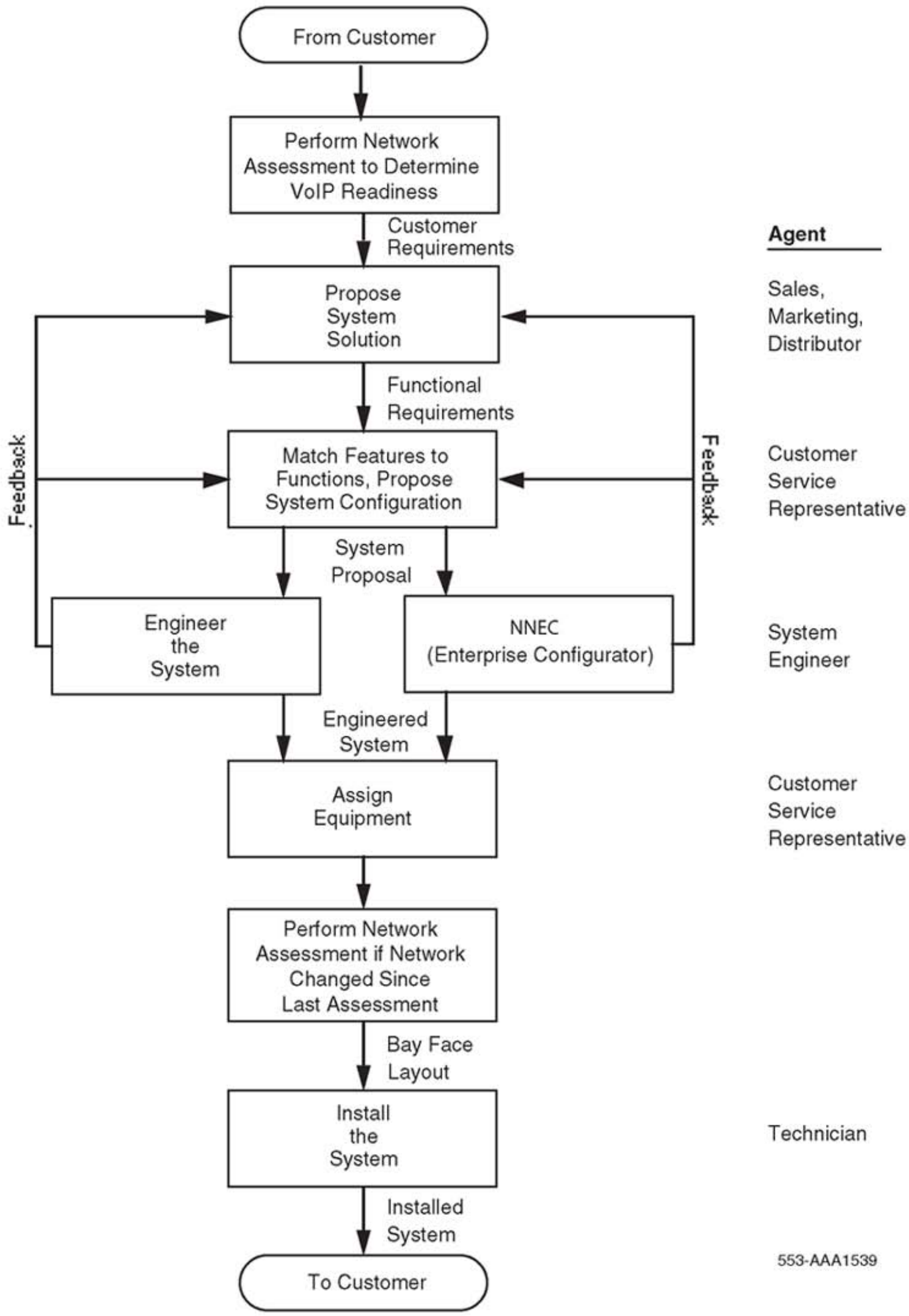

Figure 1: Engineering a new system

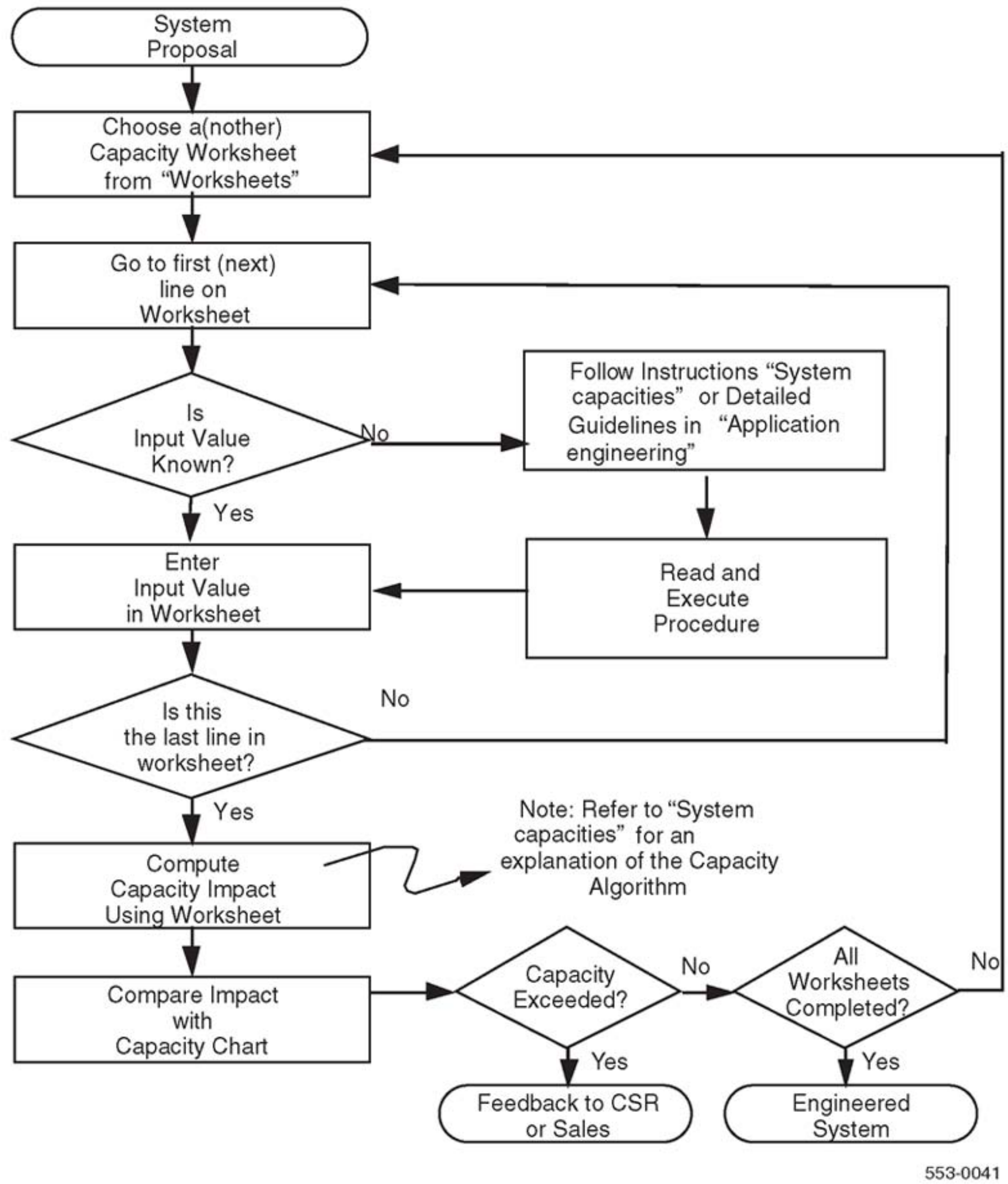

Figure 2: Engineering a new system

Communication Server 1000 task flow

This section provides a high-level task flow for the installation or upgrade of an Avaya Communication Server 1000 (Avaya CS 1000) system. The task flow indicates the recommended sequence of events to follow when configuring a system and provides the publication 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 1000M and Meridian 1 Large System Installation and Commissioning, NN43021-310*
- *Avaya Communication Server 1000M and Meridian 1 Large System Upgrades Overview, NN43021-458*

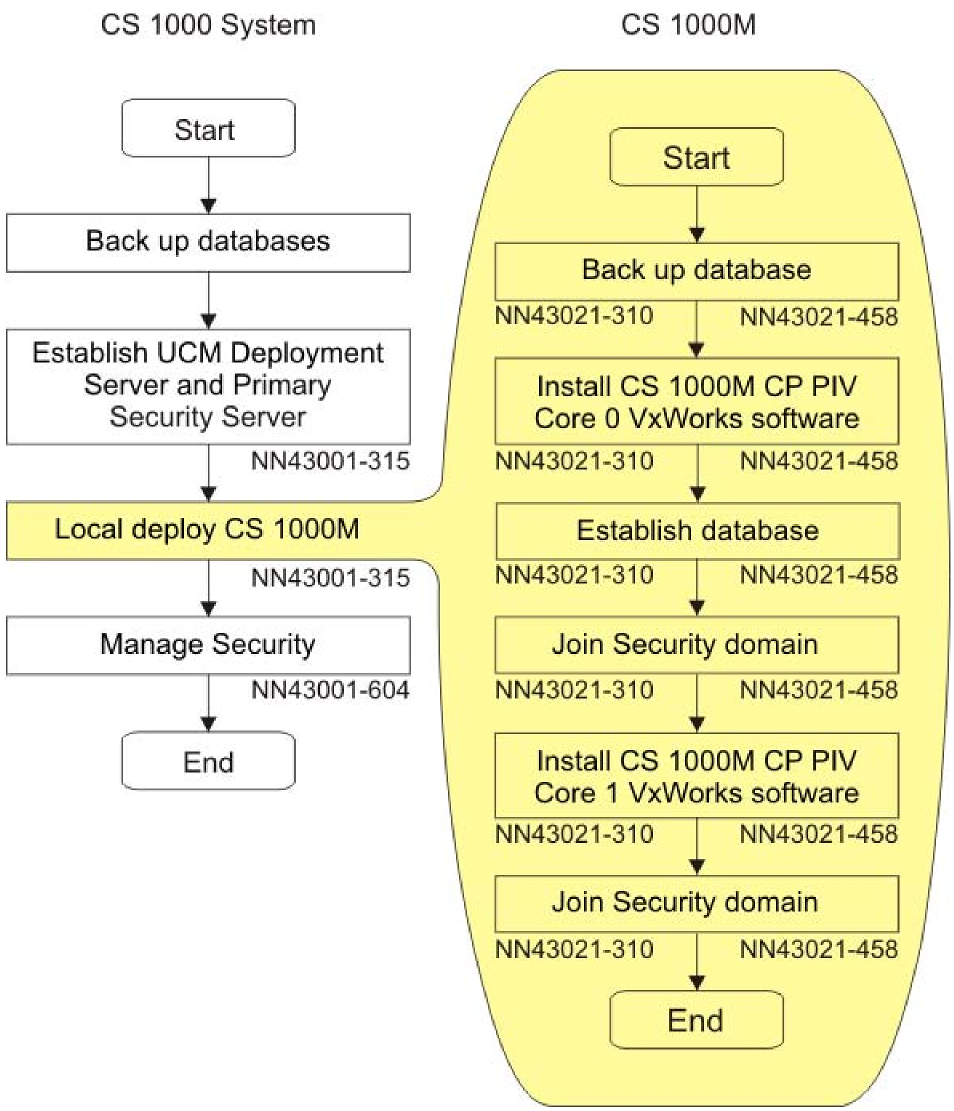

Figure 3: Avaya CS 1000M task flow

Other resources

This section briefly describes the tools available to assist the site engineer, sales person, and/or customer in engineering the switch. Differences between the tools, their platforms, and implementation and usage are described.

Enterprise Configurator

The Enterprise Configurator (EC) is a global engineering and quotation tool, available in both stand-alone and web-based versions. For users in North America and CALA, it replaces Meridian Configurator and 1-Up. For users in EMEA countries, it replaces NetPrice.

EC provides a simple "needs-based" provisioning model that allows for easy configuring and quoting. It supports Communication Server 1000M and Meridian 1 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 Communication Server 1000M and Meridian 1 configurations. Graphics depict the engineered platform, showing how the shelves are populated with various cards as well as loop assignments.

EC runs on the user's Windows or Mac personal computer. It uses standard browser and Microsoft Office applications. For details on computer system requirements and for user instructions, see the Avaya web site.

EC implements the algorithms specified in this document for real time, memory, and physical capacities. It is the official tool for determining whether a proposed configuration will meet the customer's capacity requirements.

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

Chapter 4: Data network planning for VoIP

Contents

This chapter contains the following topics:

Introduction on page 21

Data network planning for VoIP on page 28

100BaseTx IP connectivity on page 30

Introduction

Warning:

Before a Large 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 Large System.

These requirements are critical to the system Quality of Service (QOS). For information about network requirements, see *Avaya Converging the Data Network with VoIP Fundamentals, (NN43001-260)* .

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 connections
    - jitter
    - bandwidth
    - LAN recommendations
- basic data network requirements for IP Phones
    - bandwidth
- power requirements for IP Phones

Evaluating the existing data infrastructure

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

To evaluate voice performance requirements, review device inventory, network design, and baseline information. Links and devices must have sufficient capacity to support additional voice traffic. It may be necessary to upgrade links that have high peak or busy hour utilization.

When analyzing the network 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 availability 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 IP Telephony.

Evaluate 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 IP Telephony and existing network requirements are met, it may be necessary to add one or more of the following: memory, bandwidth, features.

Planning deployment of a Large System on a data network

To deploy the Large System on a data network, consider the following details and see *Avaya Converging the Data Network with VoIP Fundamentals, (NN43001-260)*:

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

QOS planning

An IP network must be engineered and provisioned to achieve high voice quality performance. QOS policies must be implemented network-wide so 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 different amounts of delay, jitter, and loss at any time. This can produce problems such as 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

There are three networks in the Large System IP Telephony network design:

1. Call Server network
2. ELAN (Management LAN) subnet
3. TLAN (Voice LAN) subnet

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

100BaseTx IP connectivity

The Avaya Communication Server 1000 (Avaya CS 1000) systems support 100BaseTx IP point-to-point connectivity or campus data network connectivity. Campus data network connectivity is provided by Media Cards and MGC DSP daughterboards in the Call Server.

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

Campus network system requirements

The following campus network 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 Media Cards, which must be on the same ELAN subnet.
- 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.

For more information about basic data network and LAN requirements for Call Server connections, see *Avaya Converging the Data Network with VoIP Fundamentals, (NN43001-260)*

- 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 in the point-to-point configuration. Do not exceed the specified round trip delay parameters.

Chapter 5: Regulatory information

Contents

This chapter contains the following topics:

System approval on page 33

Electromagnetic compatibility on page 34

Notice for United States installations on page 35

Notice for Canadian installations on page 37

Canadian and US network connections on page 38

NT4N49AA 4-Feed PDU on page 120

System approval

The Large 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 may 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 Communication Server 1000 (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 may 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 Avaya CS 1000M, Communication Server 1000S, and Communication Server 1000E systems, as well as the Meridian, Succession 1000, and CSE 1000 systems.

In order to comply with Regulatory Requirements, all components, (Including doors, covers, side panels, inter-cabinet spacers, ferrites, EMI gaskets, shielded cables, etc.,) must be in place and operative while the system is in service.

Electromagnetic compatibility

Caution:

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

If a Signaling Server is added to a previously CISPR Class B system (previously used in some specific countries), the system is now compliant to Class A.

Table 1: EMC specifications for Class A devices on page 34 lists the Electromagnetic Compatibility (EMC) specifications for the system.

Table 1: EMC specifications for Class A devices

Jurisdiction	Standard	Description
United States	FCC CFR 47 Part 15	FCC Rules for Radio Frequency Devices (see Note 1)
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

Jurisdiction	Standard	Description
Taiwan	CNS 13438	Limits and methods of measurement of radio disturbance characteristics of information technology equipment
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, may 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 will be required to correct the interference at his own expense." The user should not make changes or modifications not expressly approved by Avaya. Any such changes could void the user's authority to operate the equipment.		
EN 55022/CISPR 22 statement: "WARNING This is a class A product. In a domestic environment this product may cause radio interference in which case the user may 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 the pedestal unit in each switching equipment column. 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 may 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 will 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 2: 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 2: 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 2: Network connection specifications

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
MTS/WATS					
2-Wire, LSA, L-S (2-Wire, Local Switched Access, Loop-Start)	02LS2	9.0F	2.7A	RJ21X CA21X*	NT8D14
2-Wire, LSA, G-S	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.

AC-powered CS 1000M systems and Meridian 1 Large Systems are not designed to meet the above specifications and are not approved for sale in Europe, Middle East, and Africa regions.

Class 1 LED device

Chapter 6: System equipment

Contents

This chapter contains the following topics:

Introduction on page 41

Universal Equipment Modules on page 42

Common Equipment (Core) on page 48

Signaling Server on page 52

Network equipment on page 54

Peripheral Equipment on page 65

Terminal equipment on page 70

Power equipment on page 72

Ongoing configuration on page 75

Introduction

This section gives a high-level description of system architecture, emphasizing components of the Avaya Communication Server 1000M (Avaya CS 1000M) and Meridian 1 that have capacity limitations or impacts. The hardware of these systems is divided into six functional areas:

1. Common Equipment (Core) – Provides the processor control, software execution, and memory functions of the system.
2. Common Equipment (Network) – Performs switching functions between the processor and Peripheral Equipment cards.
3. Peripheral Equipment – Provides the interface between the network and connected devices, including terminal equipment and trunks.
4. Terminal equipment – Includes telephones and attendant consoles (and may include equipment such as data terminals, printers, and modems).

5. Power equipment – Provides the electrical voltages required for system operation, and cooling and sensor equipment for system protection.
6. Auxiliary equipment – Includes separate computing platforms that provide additional functionality which interfaces with and sometimes controls the activities of the switch's main processor.

As shown in Figure 4: Basic system architecture on page 42, the network interface function is generally considered a subset of the Common Equipment functions.

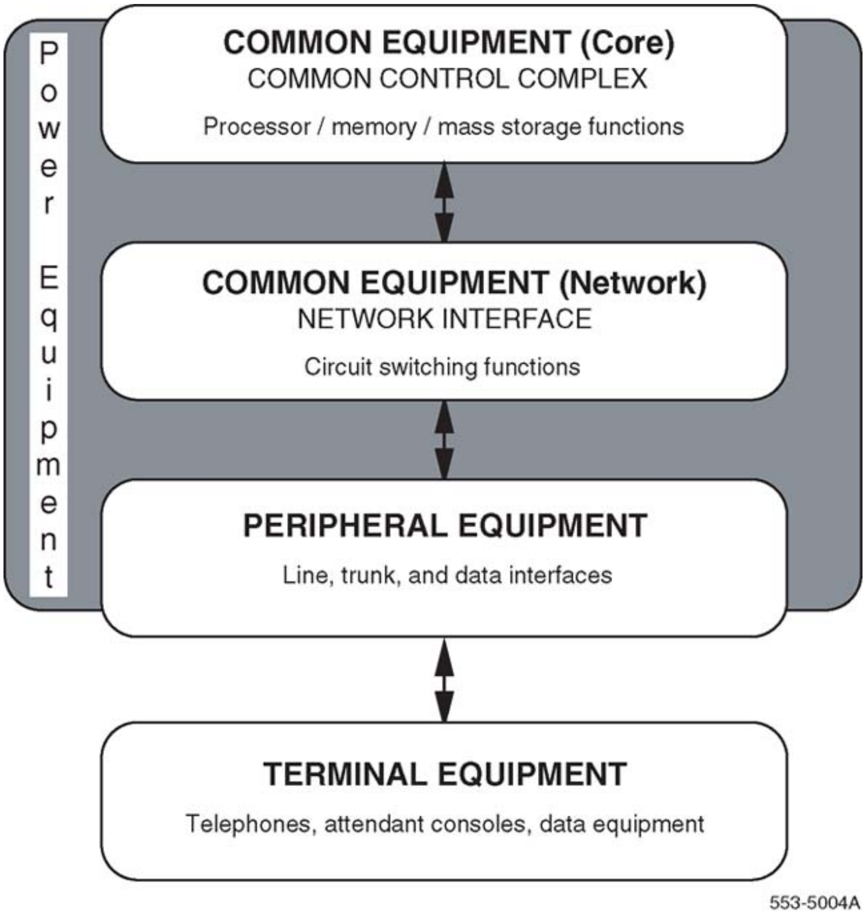

Figure 4: Basic system architecture

This section provides guidelines for system configuration. The worksheets referenced in this section can be found in Worksheets on page 381.

Universal Equipment Modules

Universal Equipment Modules (UEM) are the building blocks of the communications system. Each UEM is a self-contained unit with power, a card cage, I/O panels, and cable routing channels. It is a generic case containing sets of equipment used in system operations (see Figure 5: Universal Equipment Modules on page 44).

UEMs are stacked in columns

UEMs are stacked in columns, up to four modules high. Within a column, the levels are referred to as tiers. The UEMs are numbered 0 to 3 from the bottom up (see Figure 5: Universal Equipment Modules on page 44). Cables connect cards in the same module, between two modules, and between cards and the I/O panel in the same module.

Column components

Each column contains a pedestal base, a top cap, and up to four modules.

Pedestals

Each column sits on a pedestal. The pedestal contains power, cooling, and monitoring equipment.

- A Power Distribution Unit (PDU) in the back of the pedestal supplies either AC or DC power to the column.
- A System Monitor checks the column's cooling and power systems.
- A blower unit (accessible from the front of the pedestal) forces air up through the modules to cool the circuit cards.

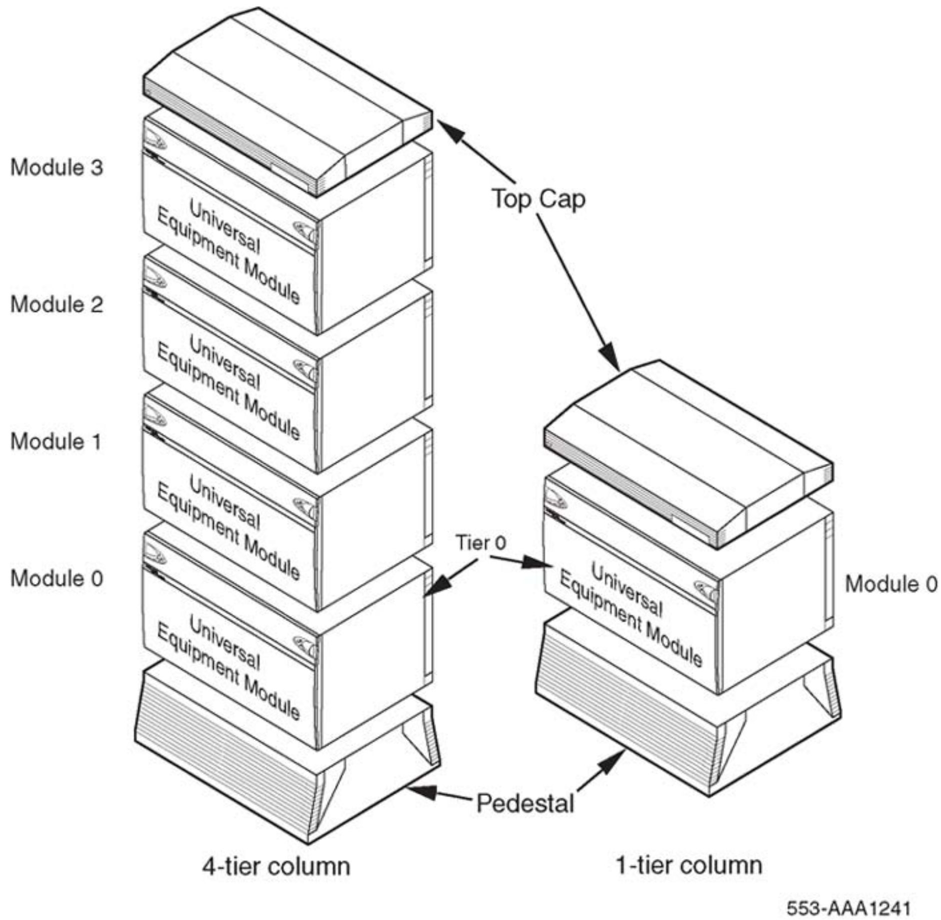

Figure 5: Universal Equipment Modules

Top Caps

A top cap is mounted on the top module of each column. It contains:

- Air exhaust grills in the cap that release air from the blowers in the pedestal.
- A heat sensor that monitors the temperature of the column.
- A red LED in the front of the cap's exhaust grill that lights if the system overheats or if a power outage occurs.
- Ladder racks for routing cables can also be fitted to the top caps.

Modules

Up to four modules can be included in a column. The modules can include:

- NT4N41 CompactPCI® (cCPI) Core/Network Module – required for all Large Systems
- NT8D35 Network Module – required for Meridian 1 Option 81C and Avaya CS 1000M MG
- NT8D37 Intelligent Peripheral Equipment (IPE) Module – required for all Large Systems

Columns are grouped in rows

A system can have one column or multiple columns. Columns are attached in rows. Column 0 is always the column containing the "Core/Net 0" module. Column 1 is placed to the left of Column 0 and ALWAYS contains the "Core/Net 1" module.

Column 0 and Column 1 are placed at the far left of the row (front view). Column numbering continues to the right of Core 0 (see Figure 6: Example of Large System column row on page 45).

Additional rows are configured with the lowest numbered column on the far left and the highest numbered column on the far right (front view).

For compliance with electromagnetic interference/radio frequency interference (EMI/RFI) standards, spacer kits are provided to interconnect the columns in a multiple-column system.

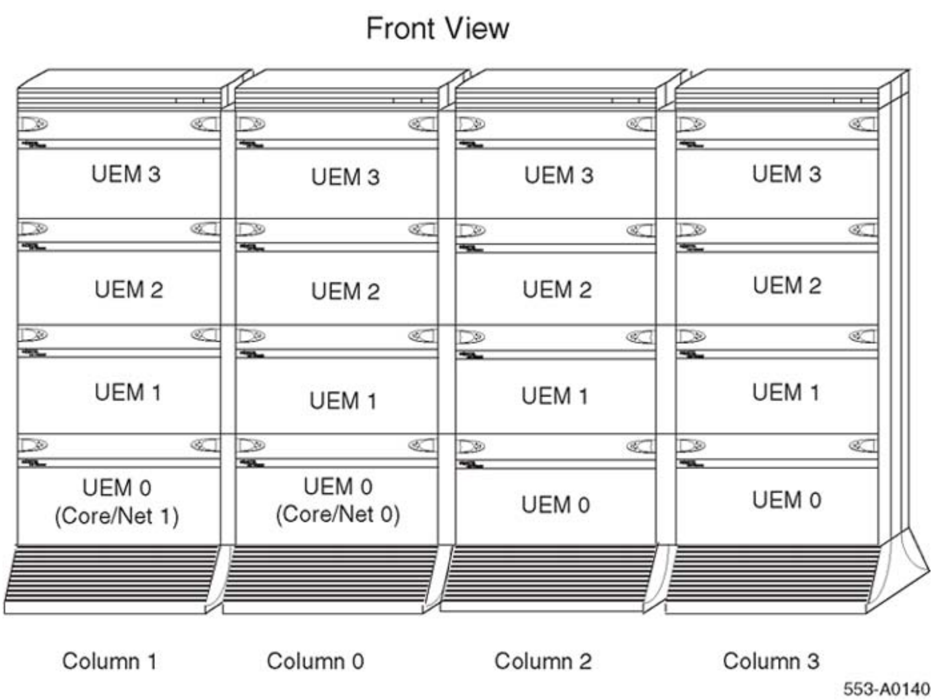

Figure 6: Example of Large System column row

Modules containing the Core Processor (CP) equipment should always be placed in the first two tiers of system columns.

UEMs are identified by function

Each UEM contains a specialized set of equipment to digitalize, process, and route phone calls and voice messages (see Figure 7: UEMs identified by function on page 46).

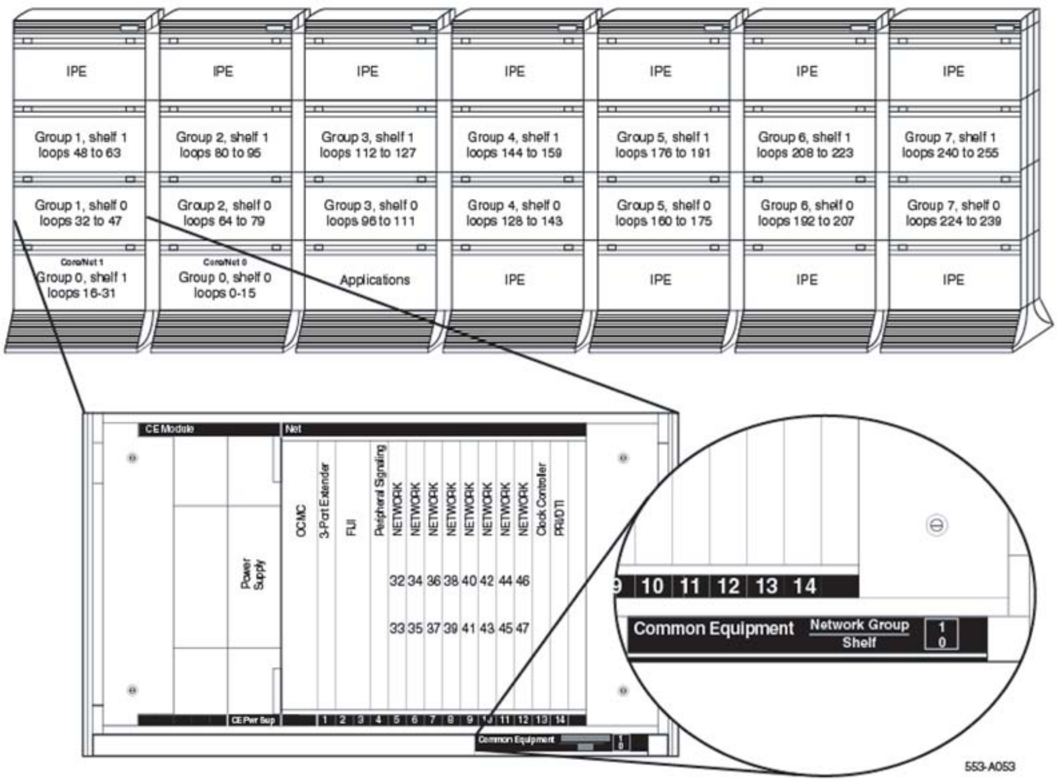

Figure 7: UEMs identified by function

Card cage

Inside each UEM is a metal card cage. This card cage holds the circuit cards, power card, and related equipment for that module. UEMs are named for the function of that card cage.

Card cages are bolted inside the UEM case. Card cages can be removed and replaced for repairs or upgrades.

Core/Network module

Meridian 1 Large Systems feature the NT4N41 Core/Network module. The Core/Network module provides a unified hardware platform for single group and multigroup configurations. The NT4N41 Core/Network module supports:

- an integrated cPCI shelf
- NT4N48 System Utility card that incorporates the functionality of the System Utility Transition card, LCD display, and the security device holder
- a fanout panel (see Figure 8: NT4N41 Core/Network shelf fanout panel (backplane) on page 48) to provide connectivity to the network shelf
- upgrades from single group to multigroup configurations (requiring a new keycode file and any additional hardware necessary for a multigroup system)

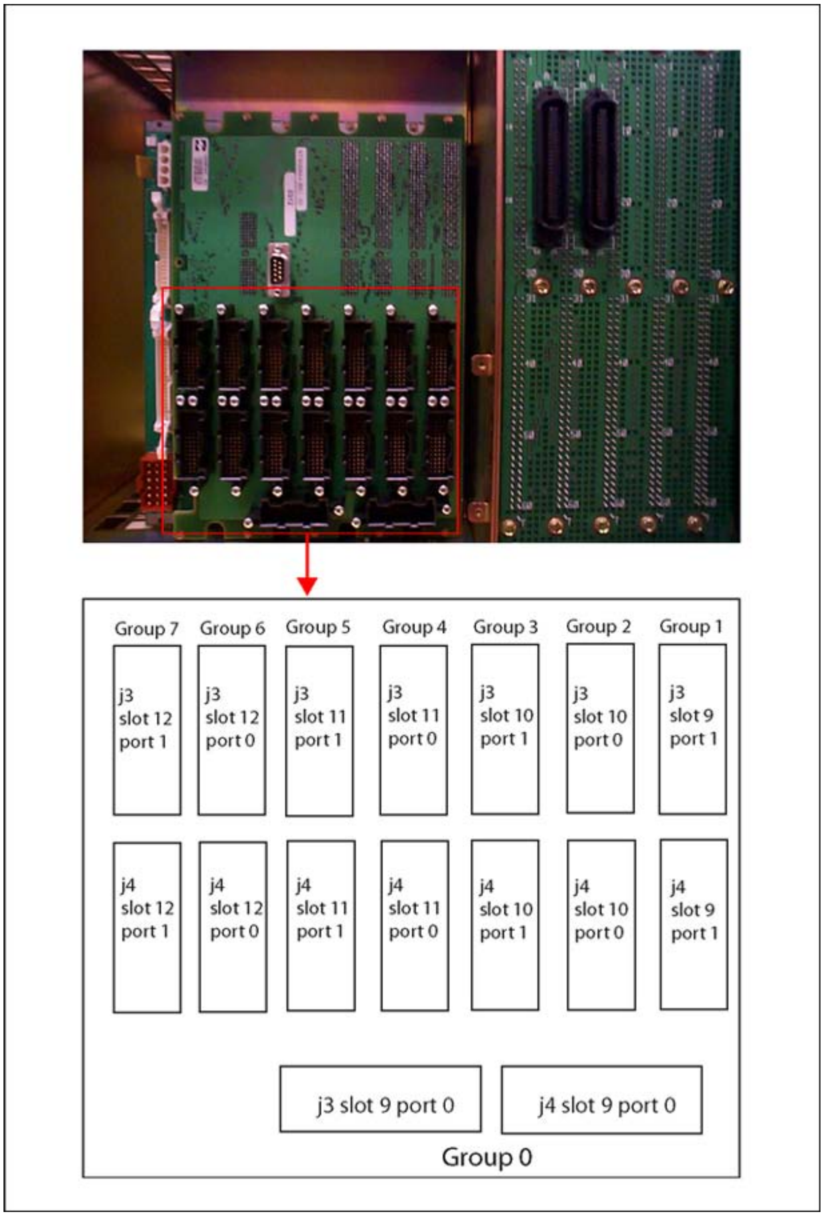

Figure 8: NT4N41 Core/Network shelf fanout panel (backplane)

Common Equipment (Core)

The central processor is the common control complex of any system. It executes the sequences that process voice and data connections, monitor call activity, and perform system administration and maintenance.

The processor communicates with the network interface over a common control bus that carries the flow of information.

The common control complex consists of:

- cPCI-based design compatible with the CP PIV chassis
- Intel Pentium IV processor
- Two Compact Flash (CF) sockets: one on-board and one hot swappable on the faceplate
- CP PIV processor card with 512MB of DRAM memory

Core Processor (CP)

At system power-up, stored instructions are executed by the CP to begin the process of loading programs from the system's Compact Flash (FMD) card into memory. The program's first activity is to read in the site's configuration database from the FMD. Once the system loading and initialization process is complete, the program enters its normal operational state.

During normal operation, the CP performs control and switching sequences required for call processing, system administration, and maintenance. It also processes input/output messages, which provide interfaces to auxiliary processors and the system administrator. The CP is capable of executing a limited number of these instructions in a given time period. This number depends on the processing power of the CP.

System memory

The CP PIV pack has an on-board PC BIOS stored in 1MB of Flash memory. The PC BIOS is used for initial configuration, but is then superseded by the VxWorks operating system for configuration parameters.

The CP PIV processor pack can support 1 GB of DDR DRAM memory in one of 2 DIMM memory sockets. The pack ships with 512MB.

System software and customer data is stored on board on a 1GB Compact Flash card that acts as a hard drive. The software is loaded from the Compact Flash (FMD) into DRAM memory prior to code execution.

On the NT4N39 (CP PIV), DRAM is divided into six functional areas:

1. Unprotected data store (UDATA) — holds constantly changing, unprotected data (such as call registers, call connection, and traffic data) required during call processing.
2. Protected data store (PDATA or office data) — holds protected customer-specific information (such as trunk configuration and speed call data).

3. Program store — holds call processing programs, input/output procedures, programmed features and options (such as conference and call transfer), and diagnostic and maintenance programs.
4. OS heap— an area from which features can allocate memory during run time by means of VxWorks memory allocation function calls. Heap users are features that are relatively self-contained and have taken advantage of the VxWorks C/C++ development environment. They include QSIG, message-based buffering, Taurus, MMIH, SMP.
5. Patching area.
6. Miscellaneous fixed OS requirements.

For the minimum memory requirements, see Memory size on page 194.

Input/output interfaces

With the NT6D80 Multiple purpose Serial Data Link (MSDL) Cards, a maximum of 64 I/O ports are supported (there are 4 ports per card; up to 16 cards can be configured). However, the maximum number of AML ports supported remains at 16.

Several types of I/O ports are available, each with its own unique protocol and bandwidth characteristics. The bandwidth of an I/O port may constrain the amount of information that can be exchanged over that link.

Network Modules

The modules for each network group must be located together (see Figure 9: Meridian 1 Option 81C with network groups on page 51):

- The two Core/Network modules are side by side in the bottom tier (Tier 0) of Column 0 and Column 1.
- For the additional network groups, the two modules that house each full network group are one on top of the other in the middle two tiers, with the module for Shelf 0 on the bottom.
- Place the first additional network group (Group 1) in Column 1; place the next network group (Group 2) in Column 0; place additional network groups in sequence to the right of the CP columns.

Figure 9: Meridian 1 Option 81C with network groups on page 51 shows a Meridian 1 Option 81C. This Large System provides the first network group (Group 0) in the two Core/Network Modules.

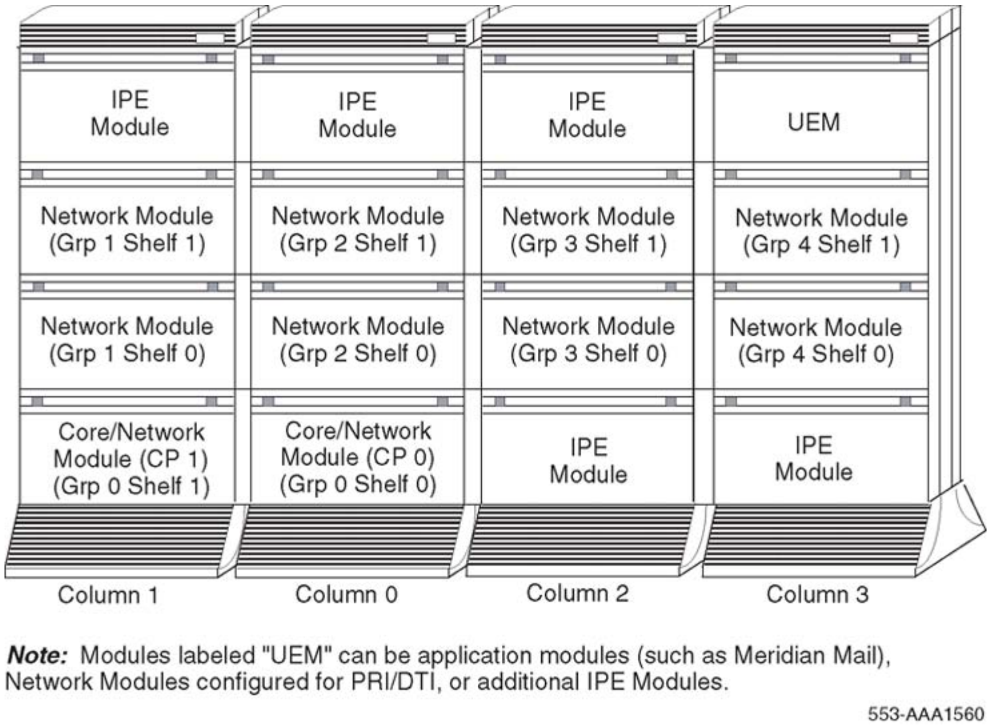

Figure 9: Meridian 1 Option 81C with network groups

Fiber Network Fabric

Fiber Network Fabric uses Fiber Junctor Interface (FIJI) cards connected with fiber-optic cable to form a Dual Ring Fiber Network. This network provides complete nonblocking communication between up to eight network groups, eliminating the incidence of busy signals for calls switched between groups.

IPE Modules

The distance allowed between a network card and the Peripheral Equipment module it serves is limited to a maximum network cable length of 13.7 m (45 ft). A Peripheral Equipment module can be placed anywhere in the system, as long as it is within the range of the network cable. For more information about IPE modules, see NT8D37 Intelligent Peripheral Equipment Module on page 81.

You can convert NT8D37 IPE modules into CS 1000E Media Gateways (MGX) with the NTDW20 Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card. For more information about MG XPEC, see Convert IPE modules into Media Gateways on page 83.

Signaling Server

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

The supported Signaling Server hardware platforms are Common Processor Pentium Mobile (CP PM), Common Processor Dual Core (CP DC), and Commercial off-the-shelf (COTS) Signaling Servers. Available COTS servers are IBM x306m, IBM x3350, HP DL320 G4, HP DL360 G7, and Dell R300 servers.

The Communication Server 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 Primary and Secondary Unified Communications Management (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 is used to run multiple applications, including:

- SIP/H.323 Signaling Gateways
- Terminal Proxy Server (TPS)
- Network Routing Service (NRS)
- SIP Line Gateway (SLG)
- Element Manager (EM)
- 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 a TLAN network interface. The Signaling Server communicates with the Call Server through the ELAN network interface.

COTS Signaling Servers mount in a 19-inch rack (see Figure 10: Communication Server 1000M Large System on page 53). The CP PM Signaling Server installs in an IPE shelf. The Signaling Server can be installed in a load-sharing redundant configuration for higher scalability and reliability.

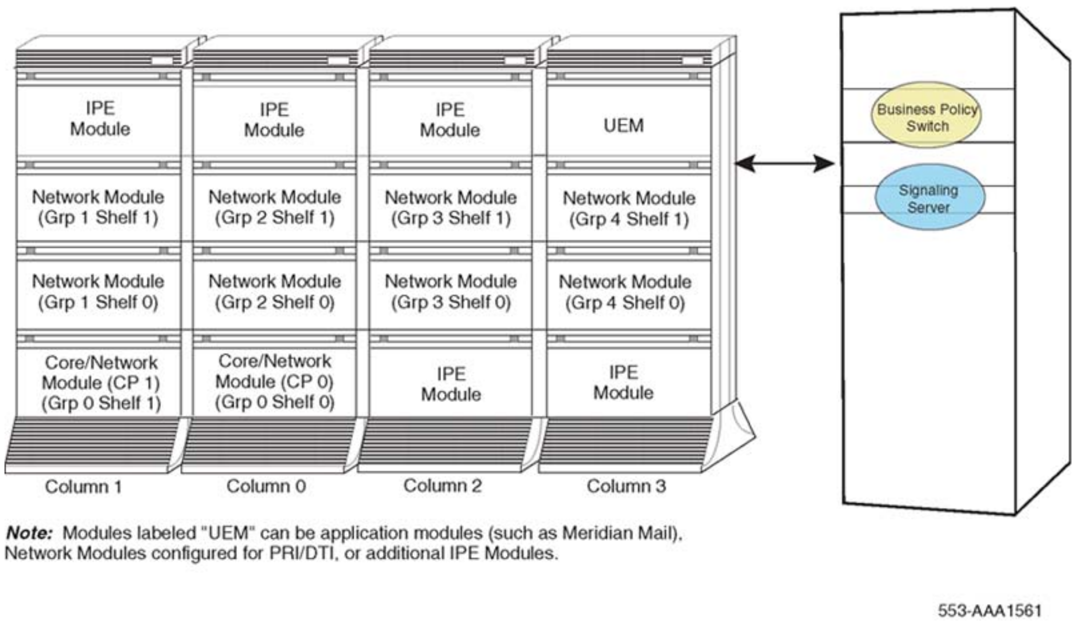

Figure 10: Communication Server 1000M Large System

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 Registrar

- Solid database component
- 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 each element's function and engineering requirements, see Table 55: Elements in Signaling Server on page 223. For detailed Signaling Server engineering rules and guidelines, see Signaling Server algorithm on page 269. 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*. For more information about the Signaling Server Linux Base, see *Avaya Linux Platform Base and Applications Installation and Commissioning, NN43001-315*

Network equipment

The network is a collection of paths over which voice and data information can be transmitted. A Communication Server 1000M or Meridian 1 network is digital, meaning that the voice and data information is encoded in digital form for transmission. These digital signals are multiplexed together on a physical entity called a loop. Each path, or channel, on a loop is identified by its timeslot, which signifies the order in which the data is placed on the loop during the multiplexing operation.

Loops transmit voice, data, and signaling information over bidirectional paths between the network and peripheral ports (that is, two channels are allocated for each conversation, one in each direction). The network is designed so that any terminal can be connected, through proper assignment of timeslots, to any other (functionally compatible) terminal on the system. The technology used is called space switching and Time Division Multiplexing (TDM).

The use of transmission channels in the switch is known as traffic. Traffic is generated by terminals (telephones and trunks). The traffic capacity of each loop or superloop is a function of the number of timeslots available and the blocking level that the user is willing to accept. Blocking is the probability that a caller will not be able to complete a call because there is no timeslot available at the particular time it is needed. The higher the traffic, the higher the blocking. A typical acceptable level of network blocking is P.01, which means 1% of all calls (1 in 100) will be blocked, on the average.

Network cards

Network cards are the physical devices that digitally transmit voice and data signals. Network switching also requires service loops (such as conference and Tone and Digit Switch [TDS] loops), which provide call progress tones and outpulsing.

The following cards provide basic network switching control.

- The NT8D04 Superloop Network card provides switching for four loops grouped together in an entity called a superloop.
- The NT5D12 Digital Trunk card provides switching for two DTI/PRI loops and takes one network slot.
- The NT5D97 Digital Trunk card provides switching for two DTI2/PRI2 loops and takes one network slot.

cCNI configuration

In the NT4N41 Core/Network Module, port 0 on the 4N65 Core to Network Interface (cCNI) card supports a half-group. This half-group does not have to be Group 0, although in a new system it is normally configured as Group 0. Communication between the cCNI and 3PE cards for Group 0 is accomplished through the backplane; no cable is required.

There are two ports on each cCNI card. Additional cCNI cards are added when additional network groups are required.

Table 3: Typical cCNI configurations on page 55 shows the default (factory) cCNI port assignments. Each cCNI card provides ports for two network groups. Connections are made from the backplane of the Core/Network modules.

Network group configuration is flexible. Any cCNI port may support any given network group. However, for ease of maintenance, associate network groups and cCNI ports in a logical sequence. See Table 3: Typical cCNI configurations on page 55 for a typical cCNI port assignment and the associated network group. Port 0 of the cCNI and the 3PE card are hardwired at the module's backplane.

Table 3: Typical cCNI configurations

cCNI card slot/port	Network group supported
cCNI 9/Port 0 cCNI 9/Port 1	Group 0 Group 1
cCNI 10/Port 0	Group 2

cCNI card slot/port	Network group supported
cCNI 10/Port 1	Group 3
cCNI 11/Port 0	Group 4
cCNI 11/Port 1	Group 5
cCNI 12/Port 0	Group 6
cCNI 12/Port 1	Group 7
You do not have to configure both ports on a cCNI card.	

The NT4N41 Core/Network Module is also used in the Communication Server 1000M SG and Meridian 1 Option 61C. Again, port 0 is dedicated to Group 0, and the cCNI card must be installed in slot 9. Port 1 is not used because the Communication Server 1000M SG and Meridian 1 Option 61C are single-group systems.

Network configuration

Network switching cards digitally transmit voice and data signals. Network switching also requires service loops (such as conference and TDS loops), which provide call progress tones and outpulsing. The NT8D04 Superloop Network card provides four loops per card. These are grouped together in an entity called a superloop.

On most systems, network loops are organized into groups.

- A single-group system (Communication Server 1000M SG) provides up to 32 loops.
- A multiple-group system (Communication Server 1000M MG) with IGS provides up to 160 loops.
- A multiple-group system (Communication Server 1000M MG) with the Fiber Network Fabric (FNF) feature provides up to 256 loops.

Superloop network configurations

By combining four network loops, the superloop network card makes 120 traffic timeslots available to IPE cards. The increased bandwidth and larger pool of timeslots provided by a superloop increases network traffic capacity for each 120-timeslot bundle by 25% (at a P.01 Grade-of-Service).

The NT8D37 IPE Module is divided into segments of four card slots numbered 0-3 (see Figure 11: Superloop segments in the IPE Module on page 57). Segment 0 consists of slots 0-3, segment 1 consists of slots 4-7, segment 2 consists of slots 8-11, and segment 3 consists of slots 12-15. A superloop can be assigned from one to eight IPE segments.

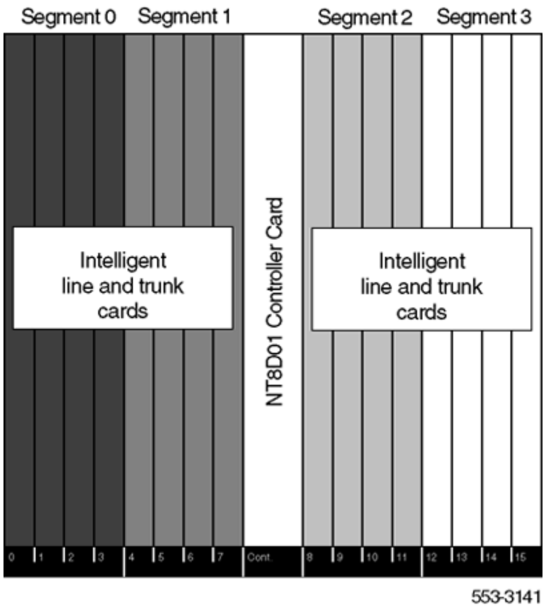

Figure 11: Superloop segments in the IPE Module

A superloop is made up of NT8D04 Superloop Network cards, NT8D01 cards, and from one to eight IPE segments. The NT8D01BC Controller-4 card interfaces with up to four superloop network cards. The NT8D01BD Controller-2 card interfaces with up to two superloop network cards.

The following superloop-to-segment configurations are supported:

- One segment per superloop on page 57
- Two segments per superloop on page 58
- Four segments per superloop on page 59
- Eight segments per superloop on page 60
- One segment per superloop/three segments per another superloop on page 61
- Two segments per superloop/six segments per another superloop on page 62

One segment per superloop

A configuration of one segment per superloop requires four superloop network cards and one NT8D01 Controller-4 card (see Figure 12: One segment per superloop on page 58).

If the segment is equipped with digital line cards that have all 16 voice and all 16 data terminal numbers (TNs) provisioned, this configuration provides a virtual nonblocking environment (120 traffic timeslots to 128 TNs).

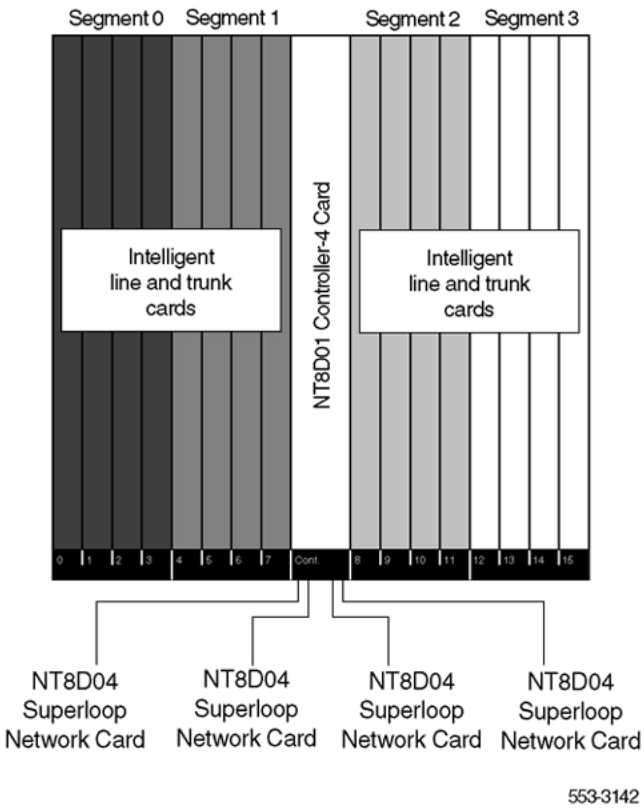

Figure 12: One segment per superloop

Two segments per superloop

A configuration of two segments per superloop requires two superloop network cards and one NT8D01 Controller-2 card (see Figure 13: Two segments per superloop on page 59).

If the segments are equipped with analog line cards and trunk cards, this configuration provides a virtual nonblocking environment (120 traffic timeslots to 32-128 TNs).

If half of the data TNs on digital line cards are enabled, this configuration still provides a low concentration of TNs to timeslots (120 traffic timeslots to 196 TNs) and a very low probability of blocking.

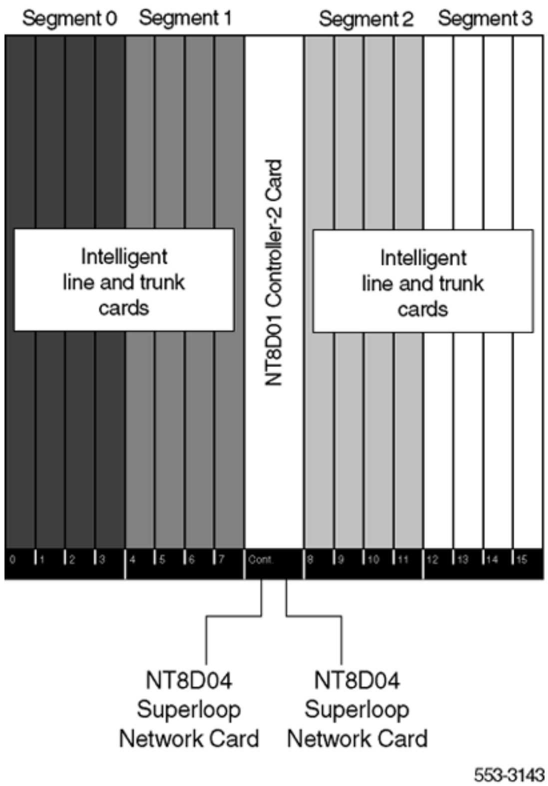

Figure 13: Two segments per superloop

Four segments per superloop

A configuration of four segments per superloop requires one superloop network card and one NT8D01 Controller-2 card (see Figure 14: Four segments per superloop on page 60).

If the segments are equipped with analog line cards and trunk cards, this configuration provides a medium concentration environment (120 traffic timeslots to 64-256 TNs).

If half of the data TNs on digital line cards are enabled, this configuration provides a concentration of 120 traffic timeslots to 384 TNs.

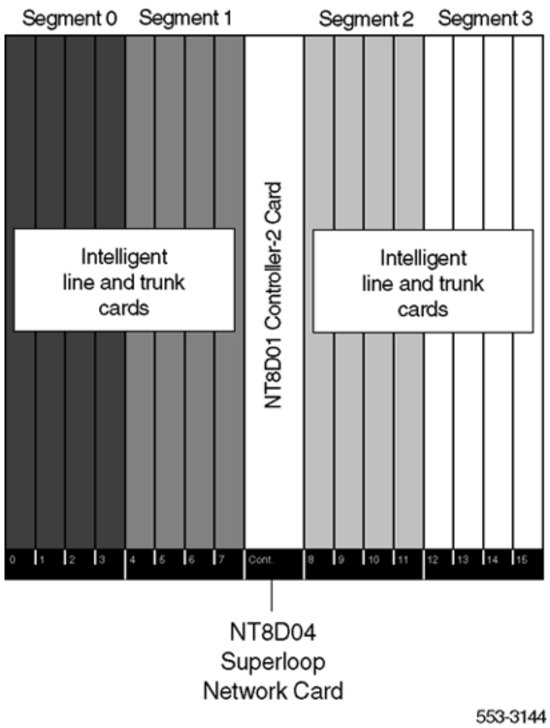

Figure 14: Four segments per superloop

Eight segments per superloop

A configuration of eight segments per superloop requires one superloop network card and two NT8D01 Controller-2 cards (see Figure 15: Eight segments per superloop on page 61).

If the segments are equipped with analog line cards and trunk cards, this configuration provides a high concentration environment (120 traffic timeslots to 128-512 TNs).

If half of the data TNs on digital line cards are enabled, this configuration provides a concentration of 120 traffic timeslots to 768 TNs.

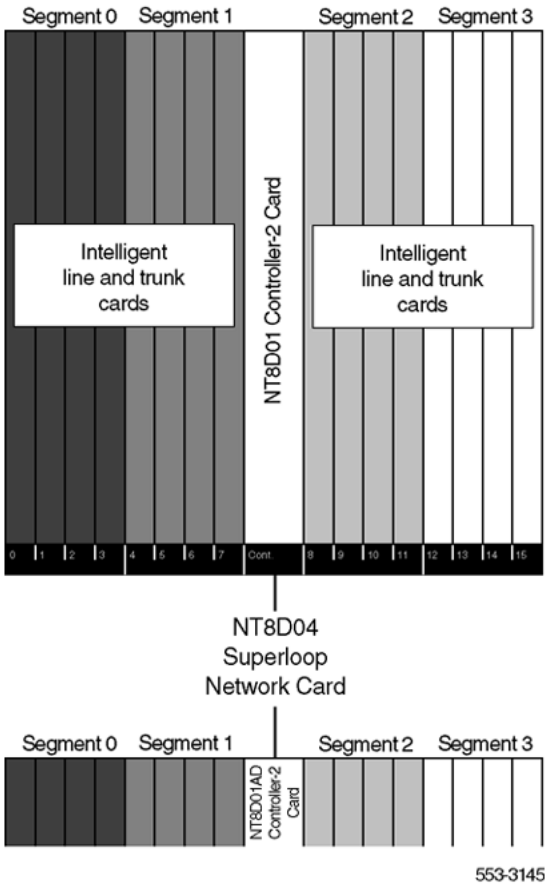

Figure 15: Eight segments per superloop

One segment per superloop/three segments per another superloop

A configuration of one segment per superloop/three segments per another superloop requires two superloop network cards and one NT8D01 Controller-2 card (see Figure 16: One segment per superloop/three segments per superloop on page 62).

This configuration provides:

- a virtual nonblocking environment (120 traffic timeslots to 128 TNs) for the single segment served by the first superloop
- a medium concentration of TNs to timeslots for the three segments assigned to the additional superloop

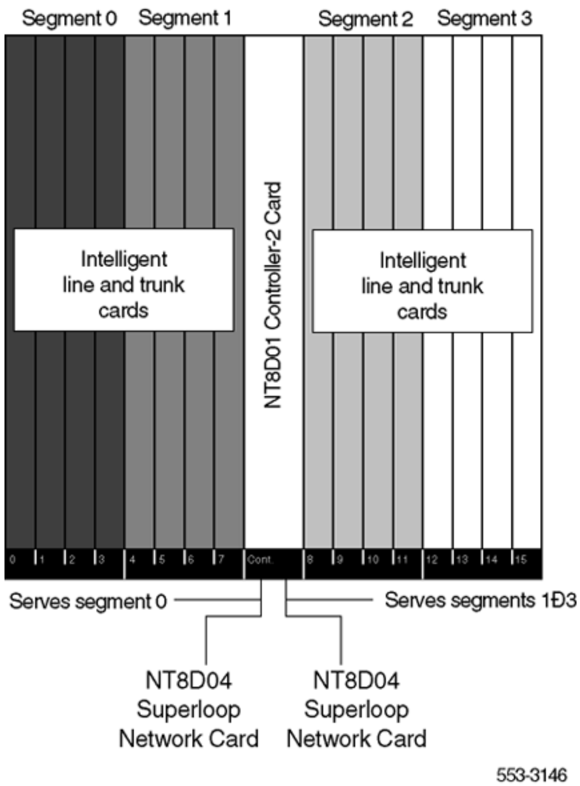

Figure 16: One segment per superloop/three segments per superloop

Two segments per superloop/six segments per another superloop

A configuration of two segments per superloop/six segments per another superloop requires two superloop network cards and two NT8D01 Controller-2 cards (see Figure 17: Two segments per superloop/six segments per superloop on page 63).

This configuration provides:

- a virtual nonblocking environment for the two segments served by the first superloop (or a very low concentration of TNs to timeslots if some data TNs are enabled)
- a medium concentration of TNs to timeslots for the six segments assigned to the additional superloop

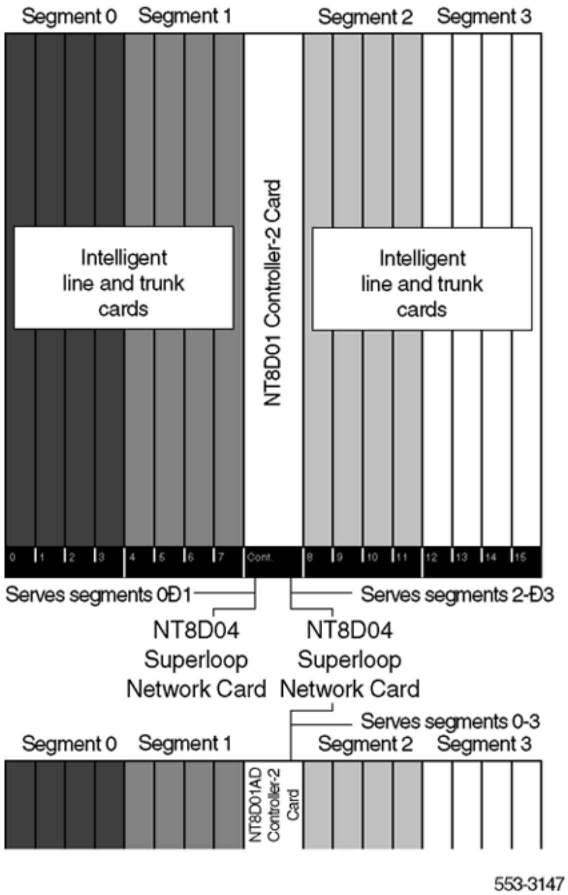

Figure 17: Two segments per superloop/six segments per superloop

Traffic configuration

The traffic distribution when considering individual customer or system traffic is shown in Figure 18: Traffic distribution on page 64.

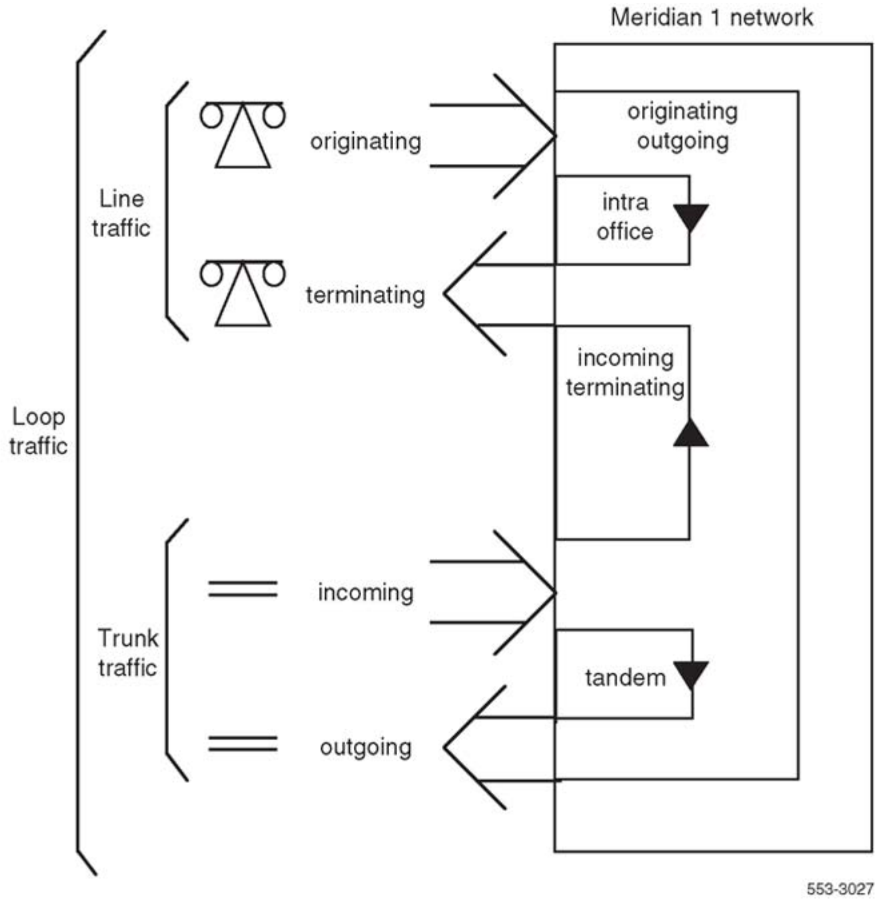

Figure 18: Traffic distribution

Network loop traffic

Typically, initial equipment is configured at an 85% utilization level to leave room for expansion. The traffic level per network loop depends on whether or not the Peripheral Equipment uses Digitone equipment:

- 3500 centi-call seconds (CCS) is the capacity of a fully loaded superloop
- 2975 CCS is 85% utilized
- Digitone traffic is a part of the capacity

Partitioning

The Communication Server 1000M and Meridian 1 Large System can be configured as a partitioned or nonpartitioned system when it serves more than one customer.

A partitioned system dedicates each customer and the customer's associated lines and trunks to actual partitioned segments of the system in terms of loops and modules. Consoles and Digitone receivers are normally spread over all loops and modules in a partitioned system.

In a nonpartitioned system, all customers, trunks, lines, consoles, and Digitone receivers are spread over all loops and modules. A nonpartitioned system provides the following advantages:

- Fewer traffic loops are required.
- Fewer Peripheral Equipment (IPE) modules and cards are required.
- System call-carrying capacity is more easily achieved and maintained.
- Customers are distributed evenly over the loops.
- Load balancing is more easily accomplished.

Network loop assignment

When assigning the loop number in systems equipped with two Network Modules, distribute the load evenly across both modules. Record the loops used in Worksheet 1:Load balancing on page 383.

Distribute the total number of IPE Modules over the total number of voice and data loops. Normally, one IPE Module is assigned to a superloop. However, one IPE Module can be assigned from one to as many as four superloops, depending on the concentration of TN-to-timeslot ratio.

Peripheral Equipment

Peripheral Equipment refers to the hardware devices that connect ports (lines and trunks) to the network (loops). Since most ports have analog voice channels and the network is digital, Peripheral Equipment cards must convert the signals received from ports from analog to digital.

A process called pulse code modulation (PCM) is used to convert analog signals to digital signals before switching is performed by the network. This conversion method samples the amplitude of the analog signal at a rate of twice the highest signal frequency, then converts the amplitude into a series of coded pulses. For telecommunications, the PCM-sampling frequency standard is 8 kHz.

Compressing-expanding (companding) PCM is a standard technique for using 8-bit words to efficiently represent the range of voice and data signals. Two standards for companding, A-Law and µ-Law, are recognized worldwide.

Intelligent Peripheral Equipment (IPE) conforms to both standards. The standard is selected through software. IPE cards are supported by NT8D04 Superloop Network card loops. IPE cards are housed in the NT8D37 IPE Module.

Intelligent Peripheral Equipment includes:

- controller cards, which provide timing and control sequences and monitoring capabilities
- analog and digital line and trunk cards, which provide interfaces to equipment outside the modules (such as telephones, data terminals, and trunks)
- lineside T1 (NT5D11) and lineside E1 (NT5D33) cards

Table 4: Intelligent Peripheral Equipment on page 66 lists the IPE cards and the number of terminations each supports.

Each equipment card contributes traffic to the network. The traffic required by a Peripheral Equipment card is the sum of the traffic generated by the ports (telephones or trunks) serviced by the card. The traffic requirements of all Peripheral Equipment cards provisioned on a particular network loop must match the traffic capacity of that loop.

Table 4: Intelligent Peripheral Equipment

Intelligent Peripheral Equipment cards	Number of terminations
Controller cards: – NT8D01 Controller-4 card – NT8D01 Controller-2 card	N/A N/A
Line cards: – NT8D02 Digital Line card – NT8D09 Analog Message Waiting Line card	16 to 32 16
Trunk cards: – NT8D14 Enhanced Universal Trunk card – NT8D15 E&M Trunk card	8 4
Terminal number (TN) density per segment is 16 to 128 TNs, with 64 to 512 TNs per IPE Module. The maximum TN density assumes all slots are equipped with NT8D02 Digital Line cards with 16 voice and 16 data TNs provisioned. A typical mix of line and trunk cards yields a nominal density of 64 TNs per segment, 256 TNs per IPE Module.	

IPE configuration

As described in Superloop network configurations on page 56, an IPE Module is divided into segments of four card slots that are assigned to superloops. A superloop combines four regular network loops to make 120 traffic timeslots available to the IPE cards. There can be from one to eight segments in a superloop, in a number of configurations. Each configuration is selected based on system traffic requirements and the specific IPE cards used.

Preferably, a superloop should be configured to serve an even number of segments. Assign full traffic and IPE cards to one superloop before assigning the next superloop. However, there may be empty IPE slots associated with a superloop if the superloop is not assigned to exact multiples of eight cards. As the system grows, more IPE cards can be added to that superloop.

The total number of ringing generators required in a system can be minimized by consolidating analog line cards in as few IPE Modules as possible. However, for traffic and reliability purposes, the analog line cards must not fill more than three-fourths of the IPE Module.

Media Cards should be configured in IPE segments engineered to be nonblocking. Avaya CallPilot should be configured in IPE segments engineered to be nonblocking.

Distributing Media Cards

Distribute a maximum of 3 Media Cards (32-port) per superloop (3 × 32 = 96, which leaves 24 timeslots for another card).

Distribute a maximum of 15 Media Cards (8-port) per superloop (15 × 8 = 120, leaving no other circuits available).

Distributing IPE cards

Use Worksheet 2:Circuit card distribution on page 384 to determine the total number of each type of IPE card (line, trunk, Digitone receiver [DTR]) for each IPE Module.

Use Worksheet 3:Multiple appearance group assignments on page 385 and Worksheet 4:Station load balancing on page 386 to determine the number of multiple appearance groups (MAGs) assigned to each loop (use Worksheet 5:Multiple appearance group record on page 387 as a MAG record sheet). Distribute MAGs evenly over all the loops.

Do not assign MAGs that call each other frequently to the same loop; assign them to the same network group to reduce intergroup calls in multiple network group systems. If possible, avoid MAGs of more than ten.

Within a multiple network group system, assign users that call each other frequently to the same network group. Similarly, assign trunk groups that are used primarily by certain groups of users to the same network group as those users.

Card slot priority

Input messages from card slots 0 and 1 in each IPE Module are directed to a high-priority input buffer. The input messages from the remaining slots are directed to a low-priority input buffer. To minimize input buffer delay on signals from devices in high-priority card slots, the system processes the low-priority input buffer only when the high-priority buffer and 500-type telephone output buffers are empty. This mechanism is important only for types of trunks that require critical timing.

Class of Service priority

Selected telephones and trunks can be assigned a high-priority Class of Service that allows their requests for dial tone to be processed first. The fewer the telephones and trunks assigned as high priority, the better the service will be during heavy load conditions.

Card slot assignment for trunks

The recommended card slot assignment for trunks is as follows:

- Always assign automatic inward and outward dial trunks to card slots 0 and 1.
- If possible, assign delay dial, wink start, and DTMF-type trunks to a high-priority card slot. Other types of trunks can be assigned to high-priority card slots to avoid glare, but can also be assigned to low-priority card slots (2 through 10).
- To minimize the number of high-priority input messages during pulsing, do not assign trunks using 10 or 20 pps (incoming) to a high-priority card slot unless necessary.

Card slot assignment for attendant consoles

Do not assign attendant consoles to a high-priority card slot. Too many high-priority messages from attendant consoles assigned to these card slots can result in delays in output messages to attendant consoles, telephones, and trunks. Always assign attendant consoles to card slots 2 through 10. Do not assign a large number of attendant consoles to the same network loop, since buffer overflow may result.

Card slot assignment for analog (500/2500-type) telephones

The 500/2500-type telephones can be assigned to any card slot. However, assigning a 500/2500-type telephone to a high-priority card slot can cause input messages to delay output buffer processing during pulsing.

Card slot assignment for Media Cards

Media Cards can be assigned to any slot. The slot should be in a nonblocking segment. For the cabinet and superloop capacities for Media Cards, see Media Cards on page 189.

Card slot assignment for CallPilot MGate/CallPilot 201i

CallPilot MGate or CallPilot 201i can be assigned to any slot. The slot should be in a blocking segment.

Assigning card slots

Use Worksheet 6:Circuit card to module assignment on page 388 to assign cards to slots in all Peripheral Equipment modules. Calculate the average load after all cards of a particular type have been assigned. Total the load and keep a running total. This method prevents the need to interchange cards at the end of the process because of load imbalance.

Assign cards in the order listed below:

1. Assign cards requiring a high-priority slot.

   For IPE Modules, both card slots 0 and 1 are reserved for high-priority signaling.
2. Assign cards for high-usage trunks, such as central office (CO) trunks.
3. Assign cards for low-usage trunks, such as paging and dictation.
4. Assign cards for attendant consoles.
5. Assign DTR cards.
6. Assign cards for telephones associated with MAGs.

7. Assign remaining cards. On a system that has a high density of Digitone telephones, assign the least number of analog line (500/2500-type telephone) cards to loops that have DTRs assigned.

   Distribute loops and Conference/TDS cards evenly across network modules and groups.
8. Calculate the total load per module.
9. Calculate the total load per loop.
10. If required, rearrange card assignments to balance the load.

Assigning terminal numbers

Once the cards are assigned, the individual units on each card can be assigned. Use Worksheet 7:Terminal number assignment on page 389 to record the terminal number (TN) assignments. TN 0000 cannot be used on superloop 0. Therefore, assign loop 0 to an NT8D17 Conference/TDS card.

Terminal equipment

Communication Server 1000Mand Meridian 1 Large Systems support a wide range of telephones, including multiple-line and single-line telephones, as well as digital telephones with key and display functions and data transmission capabilities. A range of options for attendant call processing and message center applications is also available. In addition, a number of add-on devices are available to extend and enhance the features of telephones and consoles. Add-on devices include key/lamp modules, lamp field arrays, handsets, and handsfree units.

Digital deskphones

In digital deskphones, analog-to-digital conversion takes place in the telephone itself, rather than in the associated peripheral line card. This eliminates attenuation, distortion, and noise generated over telephone lines. Signaling and control functions are also handled digitally. Time compression multiplexing (TCM) is used to integrate the voice, data, and signaling information over a single pair of telephone wires.

For applications where data communications are required, digital deskphones offer an integrated data option that provides simultaneous voice and data communications over single-pair wiring to a port on a digital line card.

The following digital telephones are supported:

- M2006 single-line telephone
- M2008/M2008HF standard business telephone
- M2216 Automatic Call Distribution (ACD) telephone
- M2317 intelligent telephone
- M2616 performance-plus telephone
- M3110 telephone
- M3310 telephone
- M3820 telephone
- Avaya 3901 Digital Deskphone
- Avaya 3902 Digital Deskphone
- Avaya 3903 Digital Deskphone
- Avaya 3904 Digital Deskphone
- Avaya 3905 Digital Deskphone
- M8000 telephone
- M8009 telephone
- M8314 telephone
- M8417 telephone

For more information about digital telephones, see *Avaya Telephones and Consoles Fundamentals, (NN43001-567)*.

IP Phones

IP Phones brings voice and data to the desktop environment and connect to the LAN through an Ethernet connection.

The Communication Server 1000M Large 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 2050 IP Softphone
- Avaya 2033 IP Conference Phone
- WLAN Handset 2210, 2211, and 2212
- IP Phone Key Expansion Module (KEM)

For more detailed information about IP Phones, see *Avaya IP Phones Fundamentals, (NN43001-368)*.

Attendant consoles

Avaya 2250 Attendant Consoles provide high-volume call processing. Indicators and a 4 × 40 liquid crystal display provide information required for processing calls and personalizing call answering. Loop keys and Incoming Call Identification (ICI) keys allow the attendant to handle calls in sequence or to prioritize answering for specific trunk groups. An optional busy lamp field provides the attendant with user status.

Meridian attendant consoles support attendant message center options. The attendant console can be connected to a personal computer to provide electronic directory, dial-by-name, and text messaging functions. All call processing features can be accessed using the computer keyboard.

The Attendant PC application software allows you to perform attendant console and call processing functions on a computer workstation using a mouse pointing device or keyboard within a Windows 95, Windows 98, Windows 2000, or Windows NT operating system environment.

Power equipment

The Communication Server 1000M and Meridian 1 provide a modular power distribution architecture.

Each column includes:

- a system monitor that provides:
    - power, cooling, and general system monitoring capabilities
    - error and status reporting down to the specific column and module
- circuit breaker protection

- a cooling system with forced air impellers that automatically adjust velocity to meet the cooling requirements of the Communication Server 1000M and Meridian 1 systems
- backup capabilities

Each module includes:

- individual power supply unit with shut-off (switch or breaker) protection
- universal quick-connect power wiring harness, which distributes input voltages and monitor signals to the power supply

All options are available in both AC-powered and DC-powered versions. The selection of an AC- or DC-powered system is determined primarily by reserve power requirements and existing power equipment at the installation site.

Although AC-powered and DC-powered systems have different internal power components, the internal architecture is virtually identical. AC- and DC-powered systems differ primarily in the external power components.

AC power

AC-powered systems require no external power components and can plug directly into commercial AC (utility) power. AC-powered systems are especially suitable for applications that do not require reserve power. They are also recommended for small to medium-sized systems that require reserve power with backup times ranging from 15 minutes to 4 hours.

If reserve power is required with an AC-powered system, an Uninterruptible Power Supply (UPS), along with its associated batteries (either internal or external to the unit), is installed in series with the AC power source (see Figure 19: External AC power architecture with reserve power on page 74). AC-powered systems that do not require long-term backup can benefit from a UPS with short-term backup because the UPS typically provides power conditioning during normal operation, as well as reserve power during short outages or blowouts.

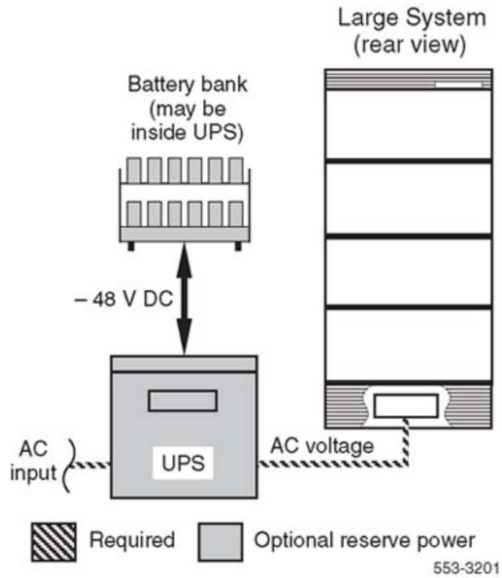

Figure 19: External AC power architecture with reserve power

DC power

DC-powered systems are available as complete systems, with external power equipment provided by Avaya. These systems can also be equipped for customer-provided external power.

DC-powered systems always require external rectifiers to convert commercial AC power into the standard –48 V DC required within the system (see Figure 20: External DC power architecture with reserve power on page 75). Batteries are generally used with DC-powered systems, as the traditional telecommunications powering method is for the rectifiers to continuously charge a bank of batteries, while the system power "floats" in parallel on the battery voltage. However, batteries are required only if reserve power is needed.

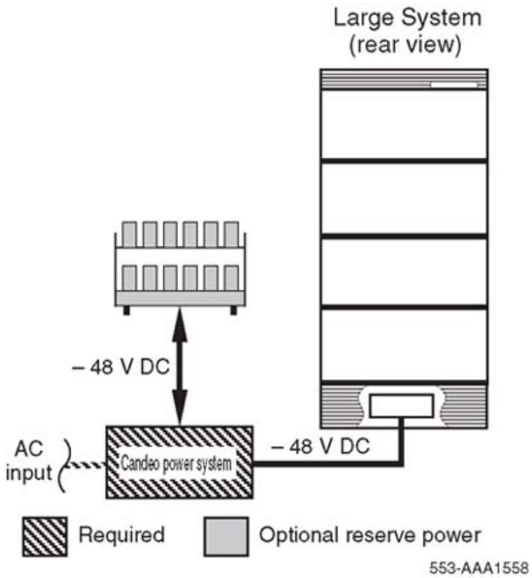

Figure 20: External DC power architecture with reserve power

Ongoing configuration

Ongoing assignment plan

Use the initial assignment records to complete an assignment plan for each equipped network loop in the system (see Worksheet 8:System assignment plan on page 390). Assignments for future trunks, MAG stations, consoles, and DTR requirements can be developed for each loop according to this profile.

Cutover study

Once the system is placed in service, perform a cutover study. Use the results of this study to update the loop profiles and create a new assignment plan. Ongoing assignments must follow the new assignment plan until the first customer busy-season trunking study. At that time, loop threshold measurements are configured so that at least one of the predominant busy hours would produce a CCS load output.

Threshold study

From the threshold study printout, update the loop profile and develop a new assignment plan. At this time, it is advisable to estimate the system capacity for growth. If the growth capacity is sufficient to last beyond the next annual threshold study, assignments can continue in accordance with the assignment plan. If the growth capacity is insufficient, plans must be made to order and install new equipment (loops or modules). The projected implementation date is generally controlled by physical capacity and tracked by total working physical terminations.

Equipment relief

When additional equipment is installed, concentrate assignments on the new loop or modules until the first threshold study. At that time, update the loop profile and develop a new loading plan. Any time a loop exceeds 2975 CCS (based on an 85% traffic level), that loop must be suspended from future assignments. If a loop encounters service problems, it must be suspended and sufficient load removed to restore service to an acceptable level.

Assignment records

The following printouts are available from the system. The printouts and the worksheets should be used to assist in maintaining assignment records.

- List of trunk route members
- List of TN blocks
- List of unused card positions
- List of unused units
- Directory number (DN) to TN matrix

For more information about obtaining and manipulating data in the system, see *Avaya Features and Services Fundamentals, (NN43001-106)* .

Chapter 7: Module configuration

Contents

This chapter contains the following topics:

Introduction on page 77

NT4N41 Core/Network Module on page 77

NT8D35 Network Module on page 80

NT8D37 Intelligent Peripheral Equipment Module on page 81

Introduction

There are three types of modules:

- NT4N41 Core/Network Module on page 77
- NT8D35 Network Module on page 80
- NT8D37 Intelligent Peripheral Equipment Module on page 81

Each type of module is available in AC-powered and DC-powered versions.

AC-powered modules generally require a module power distribution unit (MPDU) to provide circuit breakers for the power supplies. DC-powered modules do not require an MPDU because a switch on each power supply performs the same function as the MPDU circuit breakers.

The figures in this section show a typical configuration for each module. (DC power supplies are shown in these examples.)

NT4N41 Core/Network Module

This module provides common control and network interface functions in the Avaya Communication Server 1000M (Avaya CS 1000M) an Meridian 1 Large Systems. With Avaya CS 1000M MG and Meridian 1 Option 81C, two Core/Network modules are installed side by side. With Communication Server 1000M SG and Meridian 1 Option 61C, the modules are stacked or installed side by side.

The NT4N41 module contains the NT4N40 card cage, which is divided into two distinct sides: the Core side and the Net side.

Core side

The Core side of the module houses the CPU. These circuit cards process calls, manage network resources, store system memory, maintain the user database, and monitor the health of the system. These circuit cards also provide administration interfaces through a terminal, modem, or LAN.

Core cards and slot assignments are:

- slots c9-c12: NT4N65 Core to Network Interface (cCNI) card. Since each cCNI card can connect to two Network groups, each Core is connected to a minimum of two groups and a maximum of eight groups. The number of cCNI cards in a system depends on the number of Network groups in that system.

  Each new base system contains one NT4N65 cCNI card per Core/Network Module. The cCNI card is located in slot c9. A P0605337 cPCI Card Slot Filler Panel must be installed to cover slots c10 to c12 if they do not contain cCNIs.

  In the NT4N41 Core/Network Module, port 0 on the NT4N65 cCNI card in slot c9 must be configured as Group 0. Communication between the cCNI and 3PE cards for Group 0 is accomplished through the NT4N29 cable. Only one cCNI card is required for Group 0.
- slots c13-c14: P0605337 cPCI Card Slot Filler Panel.
- slot c15: NT4N48 System Utility card.
- slot CP: NT4N39AA Call Processor card (512 MByte memory).

Net side

The Net side of the module supports one or two Conference/Tone and Digit Switch (TDS) cards, one Peripheral Signaling card, one 3-Port Extender card, up to three Superloop cards, and, where slots permit, any Input/Output-type card, such as the MSDL.

Net cards and slot assignments are:

- slot 0: NT8D17 Conference/TDS card.
- slot 1: NT8D17 Conference/TDS card or NT5D12 Dual DTI/PRI 1.544 Mbps (DDP) card or NT5D97 Dual DTI2/PRI2 2.048 Mbps (DDP2) card.

- slots 2-7: NT8D04 Superloop Network (SNET) card or Input/Output (I/O)-type cards.
- slots 8-9 (Communication Server 1000M MG and Meridian 1 Option 81C): NTRB33 Fiber Junctor Interface (FIJI) card.

  Note:

  The double slot FIJI(NTRB33AF) card sits in slots 8 and 9 on the Net side of the Core/Net module whereas the single slot FIJI(NTRB33BBE5) card sits in slot 9 on the Net side of the Core/Net module.
- slots 8-9 (Communication Server 1000M SG and Meridian 1 Option 61C): NTRB53 Clock Controller in slot 9; slot 8 is spare or can be used for any I/O card, such as MSDL.
- slot 10: QPC43R Peripheral Signaling card
- slot 11: QPC441 3-Port Extender card

In Communication Server 1000M SG and Meridian 1 Option 61C systems, the NTRB53 Clock Controller card goes in the NT4N41 Core/Network Module (in slot 9). In Communication Server 1000M MG and Meridian 1 Option 81C systems, the Clock Controller goes in the NT8D35 Network Module.

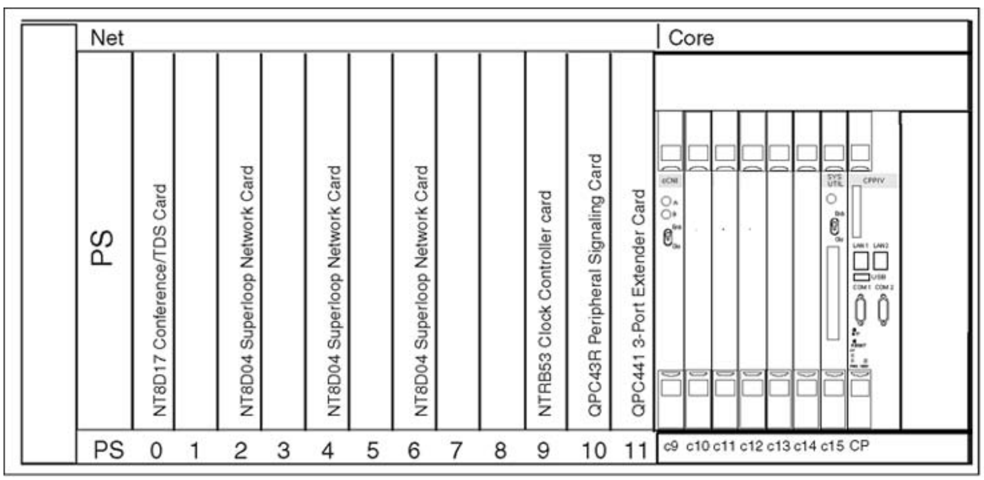

Figure 21: NT4N41 Core/Network Module (for Communication Server 1000M SG/Meridian 1 PBX 61C CP PIV)

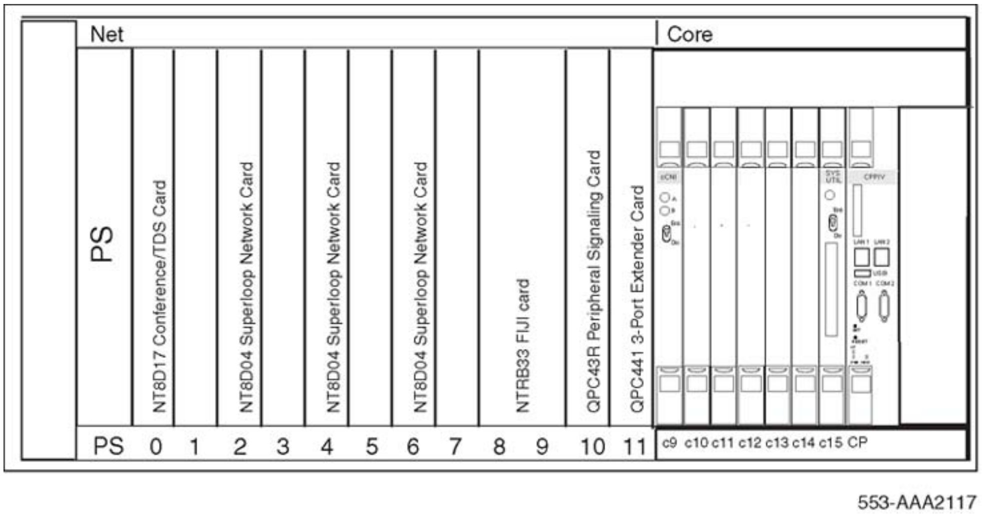

Figure 22: NT4N41 Core/Network Module (for Communication Server 1000M MG/Meridian 1 PBX 81C CP PIV)

NT8D35 Network Module

The Network Module houses up to three NT8D04 Superloop cards, one NT8D17 Conference/ TDS card, and one of the following cards residing next to the NT8D17 Conference/TDS card: NT5D12 1.544 Mbps DDP card, or NT5D97 2.048 Mbps DDP2 card.

The network cards are cabled to the Intelligent Peripheral Equipment Controller card in IPE Modules. In a typical configuration, one Conference/TDS card is configured in the module, leaving 14 voice/data loops available. Two Network Modules are required to make a full network group of 32 loops. A maximum of 16 Network Modules (8 network groups) can be configured in an FNF-based Communication Server 1000M MG or Meridian 1 Option 81C.

This module provides 15 card slots for the following network interface cards:

- between PS and slot 1: NTRE39 Optical Cable Management Card (OCMC)
- slot 1: QPC441 3PE card
- slots 2-3: NTRB33 Fiber Junctor Interface (FIJI) card

  Note:
  The double slot FIJI(NTRB33AF) card sits in slots 2 and 3 on the Network module whereas the single slot FIJI(NTRB33BBE5) card sits in slot 2 on the Network module.

- slot 4: QPC43 Peripheral Signaling card
- slots 5-12:
  - NT1P61 Fiber Superloop Network card

- NT8D04 Superloop Network card
- NT8D17 Conference/TDS card
- PRI/DTI card
- SDI-type card
- MSDL card
- MISP card
- For the Meridian 1 Option 61C, a Clock Controller is required in slot 9.

• slot 13: Clock Controller for Meridian 1 Option 81C

• slots 13-14: PRI/DTI card or SDI-type card (slot 13 only)

• slot 15: not used

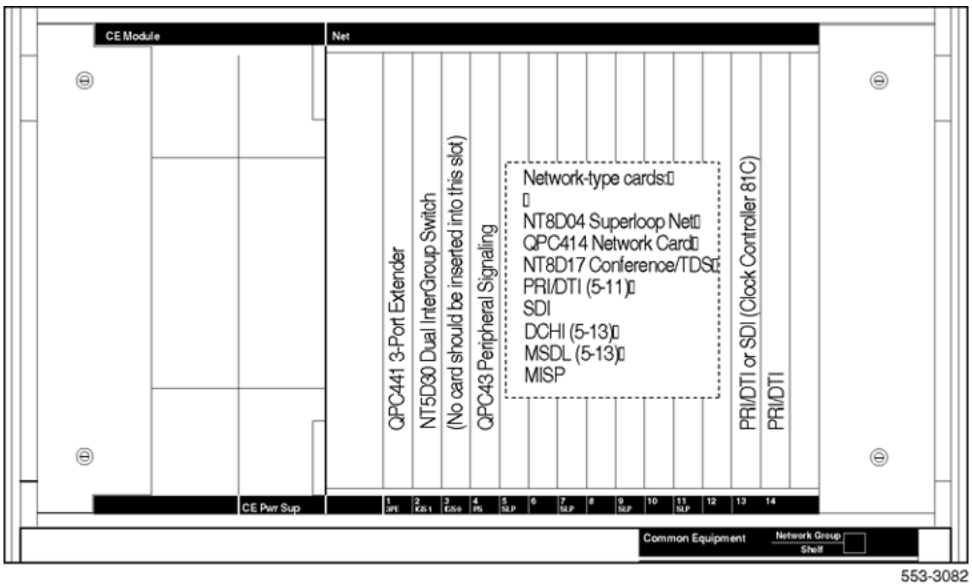

Figure 23: NT8D35 Network Module

NT8D37 Intelligent Peripheral Equipment Module

This module can be used in all systems.

The IPE Module houses one NT8D01 Controller card or one NT1P62 Fiber Peripheral Controller card and up to 16 IPE cards (such as line and trunk cards), supporting up to 512 terminal numbers (256 voice and 256 data). The controller card is cabled to the NT8D04 Superloop Network card.

The controller card must be installed in the card slot labeled Cont (for controller). The other slots can house any IPE card (see Figure 24: NT8D37 IPE Module on page 83).

When the backplane is configured for 16 cables (NT8D37 vintages BA and EC), the NT7D16 Data Access Card (DAC) can be installed in any IPE slot. If the backplane is configured for 12 cables (NT8D37 vintages AA and DC), you must install the DAC in slots 0, 4, 8, or 12 because only those slots are fully cabled for 24 pairs.

The IPE Module supports universal card slots for flexible configurations of trunk/line, application cards, and Media Cards.

Application cards

The supported application cards are:

- Meridian Integrated Recorded Announcement
- Meridian Integrated Conference Bridge
- Meridian Integrated Personal Call Director
- Meridian Integrated Voice Service
- Meridian Integrated Call Assistant
- Media Card
- CallPilot MGate

Line-side Peripheral cards

Supported line-side Peripheral cards are:

- LSI1
- LSE1

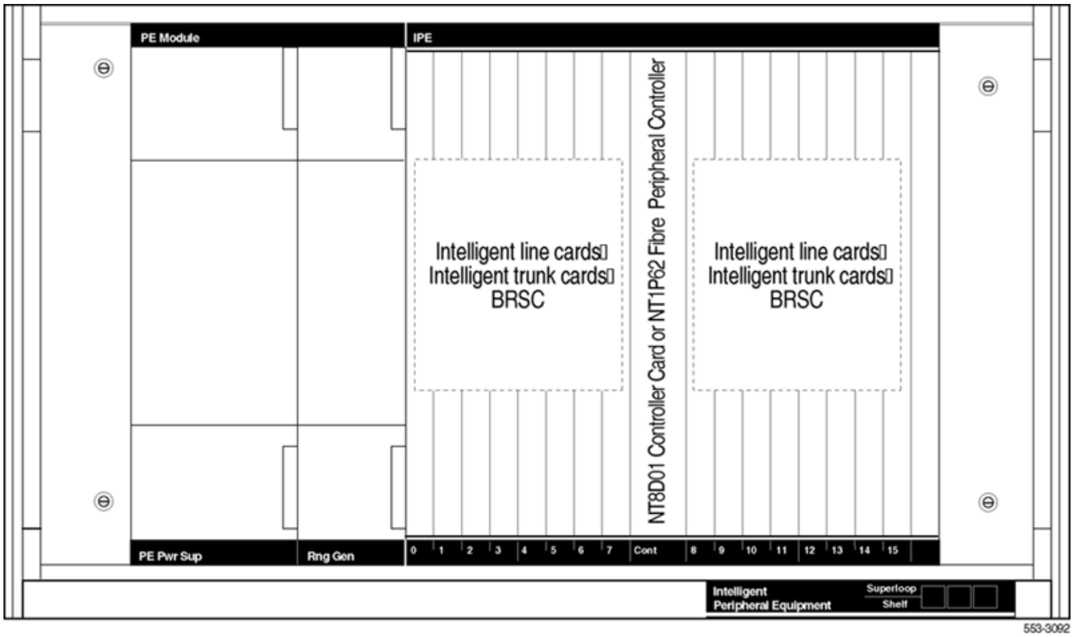

Figure 24: NT8D37 IPE Module

Convert IPE modules into Media Gateways

You can convert NT8D37 Communication Server 1000M and Meridian 1 large system IPE modules into Communication Server 1000E Media Gateways with a 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.

Media Gateway Extended Peripheral Equipment Controller (MG XPEC) card

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 provides the same hardware functions as the Media Gateway Controller (MGC) card in a traditional CS 1000E Media Gateway chassis or cabinet.

Figure 25: MG XPEC faceplate on page 84shows the faceplate of the NTDW20AAE6 MG XPEC card.

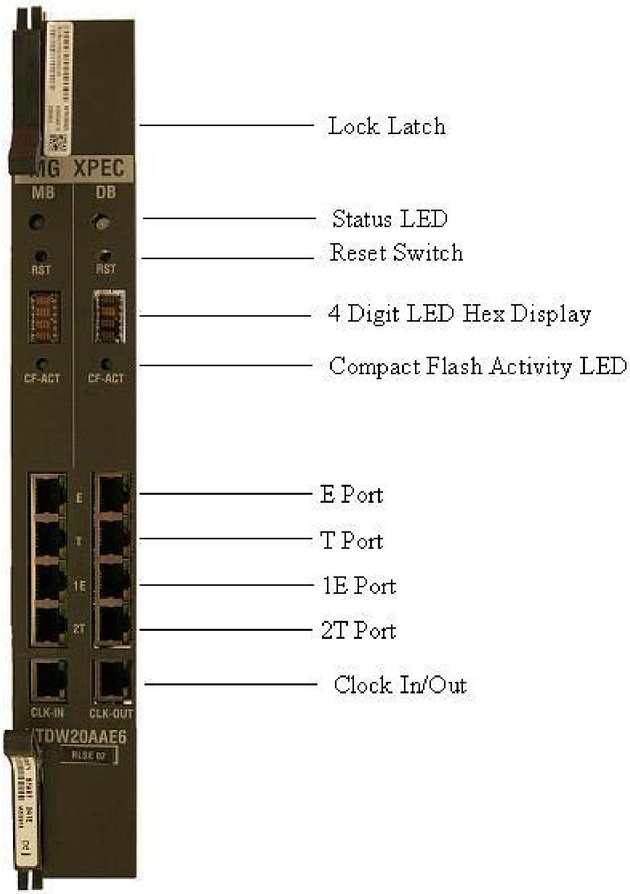

Figure 25: MG XPEC faceplate

Figure 26: MG XPEC motherboard and daughterboard on page 85 shows a MG XPEC card with the motherboard and daughterboard disassembled and offset. The MG XPEC card arrives fully assembled. This disassembled view is for information purposes only.

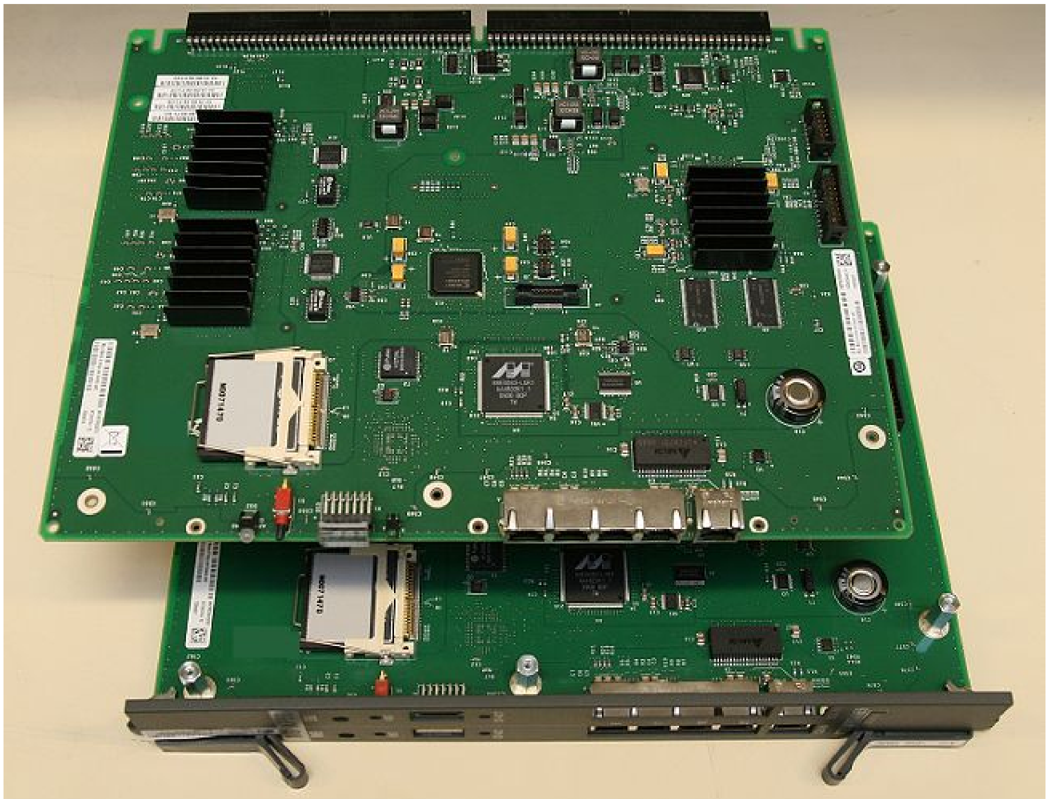

Figure 26: MG XPEC motherboard and daughterboard

The motherboard and daughterboard of the MG XPEC card provide the following features:

- Mindspeed Chagall-2 processor
- Two DSP daughterboards provide 192 DSP ports with media security
- Embedded Ethernet switch with ten ports
- Two faceplate 100BaseT ELAN ports
- Two faceplate 100BaseT TLAN ports
- Two remote TTY ports
- Faceplate four character hex display
- SDRAM and local Flash Boot-ROM
- Compact Flash (CF) card containing program store and file system
- FPGA for Avaya proprietary connections
- In-rush power controller supports hot-plugging

The MG XPEC motherboard provides the following extra features:

- Stratum 4 clock circuitry allows clock daisy chaining across several MG XPEC
- One remote TTY for Extended System Monitor (XSM)

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

Media Gateway Extended (MGX)

The MG XPEC card transforms one IPE module into two Media Gateways. Each Media Gateway registers as an MGX to the Communication Server 1000E Call Server. The MG XPEC motherboard controls the eight physical card slots (0-7) to the left of the MG XPEC card. The MG XPEC daughterboard controls the eight physical card slots (0-7) to the right of the MG XPEC card. The MG XPEC cable kit (NTDW25AAE6) includes a label to renumber slots 8 to 15 as 0 to 7 on the right of the IPE module. Logical slots 8, 9, and 10 are dedicated for the MG XPEC DSP resources. Figure 27: MGX card slots on page 86 shows the card slot numbering of an IPE module converted into two Media Gateway Extended (MGX) shelves. Each MGX shelf communicates with the Call Server through the ELAN ports on the MG XPEC card.

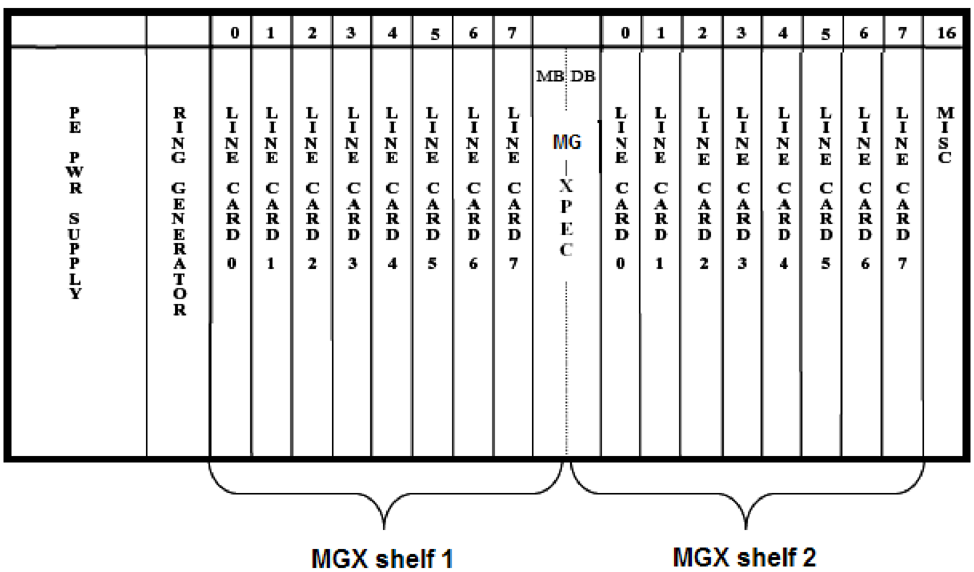

Figure 27: MGX card slots

Operating parameters

The MGX operates under the direct control of the CS 1000E Call Server. The Call Server can control up to 25 IPE modules with MG XPEC cards. A total maximum of 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 MG XPEC card provides 192 DSP ports on the motherboard and daughterboard for a total of 384 DSP ports. You can install Voice Gateway Media Cards into a MGX to provide additional DSP ports beyond the 192 port limit of the MG XPEC motherboard and daughterboard.

The MGX supports the following circuit cards and applications:

- Voice Gateway Media Cards: transcode between the IP network and digital circuit cards
- Service cards: provide services such as Music or Recorded Announcements (RAN)
- Analog interfaces to lines and trunks: support analog (500/2500-type) phones and fax, analog PSTN trunks, and external Music or RAN sources
- Analog trunk cards
- Digital line cards: support digital terminals, such as attendant consoles, M2000 and Avaya 3900 Series Digital Deskphones, and external systems that use digital line emulation, such as Avaya CallPilot Mini
- 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

CEMUX cards such as Digital Trunks are not supported in IPE modules with MG XPEC cards. The following IPE cards supported by the NT8D01 XPEC card are not supported by the NTDW20 MG XPEC card.

- NT6D70 BRI SILC Voice/Circuit/Packet Data Lines
- NT6D71 BRI UILC Voice/Circuit/Packet Data Lines
- NT6D72 BRSC Basic Rate Concentrator
- NT7D16 Data Access Line Card

Clocking connections

You can synchronize the TDM clocks of multiple IPE modules with MG XPEC cards. Use customer supplied CAT5 Ethernet cables to daisy chain the MG XPEC cards clock in and out faceplate ports. One MG XPEC card is the clock master sending a 8 Khz clock reference from the clock out port to the clock in port of the MG XPEC card in the next IPE module. You can daisy chain the clocks of up to four MG XPEC cards.

If the MG XPEC clock master fails, the next MG XPEC in the daisy chain becomes the new clock master. The other MG XPEC cards will track to the new clock master.

Cable connections

You must install a NTDW25AAE6 cable kit to provide cable connections for the MG XPEC card motherboard and daughterboard in an IPE module. Figure 28: MG XPEC cable kit on page 88 shows the contents of the MG XPEC cable kit.

Figure 28: MG XPEC cable kit

The NTDW25AAE6 MG XPEC cable kit includes the following:

- 2 NTDW26AAE6 TTY cables
- 2 Input Output knock-out panel (left and right)
- 8 CAT5e shielded Ethernet cables
- 12 RJ45 couplers
- 1 card label to renumber card slots on right side

For information about installing the MG XPEC and cable kit, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*

An optional NTDW26BAE6 XSM (Extended System Monitor) cable can connect the MG XPEC to an XSM (NT8D22). You can connect one MG XPEC to an XSM to enable monitoring of the XSM status.

Network connections

The ELAN of the MGX can reside in a separate layer 3 subnet from that of the Call Server ELAN. When connecting the MGX to the ELAN through a Layer 3 switch the connection from the Call Server to the MGX must have a round trip delay of less than 80 msec and have a packet loss of less than 0.5 % (0% recommended). You can configure routes on the ELAN interface of each MGX using Element Manager (EM).

Figure 29: Network connections with MGX dual homing on page 89shows a schematic representation of network connections for MGX dual homing.

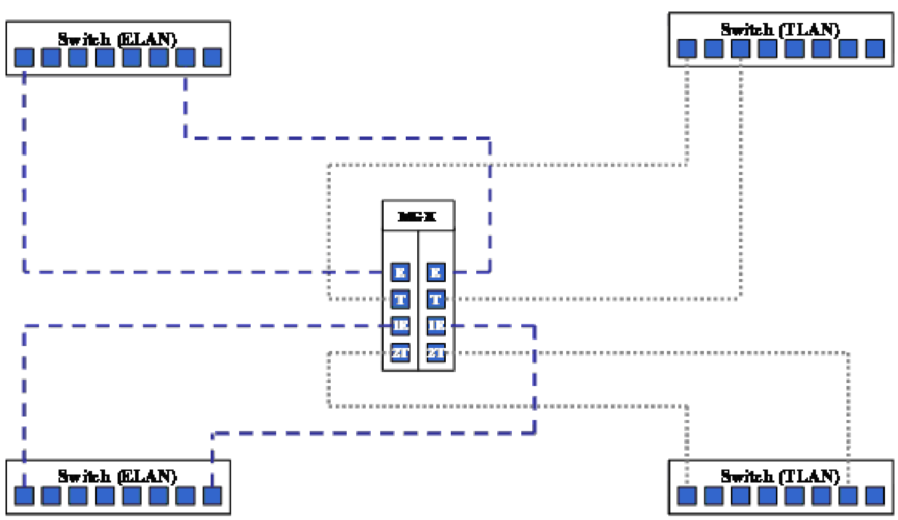

Figure 29: Network connections with MGX dual homing

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.

For more information about network cabling options, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*

Chapter 8: Power and grounding

Contents

This chapter contains the following topics:

Introduction on page 91

Grounding on page 92

Power systems on page 101

AC power on page 102

DC power on page 107

Reserve power on page 113

NT4N49AA 4-Feed PDU on page 120

Power consumption on page 128

Power requirements for IP Phones on page 129

Heat dissipation on page 129

Calculating system power requirements on page 130

System upgrades on page 131

Introduction

Large Systems can be powered by either AC power or DC power. This chapter describes:

- grounding requirements for all systems
- the power and reserve power systems that are available
- power consumption of Large System components, in order to calculate system power requirements

Grounding

Proper grounding is essential for trouble-free system operation and the safety of personnel.

Grounding recommendations for Communication Server 1000M Large Systems

To ensure electrical system grounding integrity, follow the isolated ground topology for all Avaya Communication Server 1000M (Avaya CS 1000M) Large System equipment implementations. Isolated ground provides the best method for avoiding the introduction of ground noise to the system from other external equipment.

When isolated ground topology is not possible, an alternative grounding method may be used if it provides the required Meridian 1 / Succession Single Point Ground (SPG) reference. The SPG source must be the AC Equipment Ground (ACEG) bus located inside the Avaya CS 1000M service panel. Service panel grounding facilities must be properly referenced to an acceptable AC grounding source, which provides a low noise, low impedance path.

Installations that have elected not to deploy an isolated ground methodology will be noted during Avaya system audits. Locations experiencing system operational performance difficulties attributed to ground noise or improper grounding methods will be required to rectify the issue.

Single Point Grounding

The Single Point Ground (SPG), otherwise known as the Star—IBN (Isolated Bonding Network), is the standard for the system. The SPG of a system is the point at which an IBN is bonded to ground. Physically, the system SPG is usually implemented as a copper busbar. See Figure 30: Single point grounding on page 93.

Any of the following busbars can be used as system SPG:

- Building principal ground (BPG), typically in single-floor buildings
- Floor ground bar (FGB), typically in multifloor buildings
- Dedicated SPG bar bonded to the building grounding system
- A section of the battery return (BR) bar of the power plant

The various subsystems (such as groups of frames or equipment) of an IBN system can be configured as individual SPG entities, connected in a star configuration to the system SPG (star IBN).

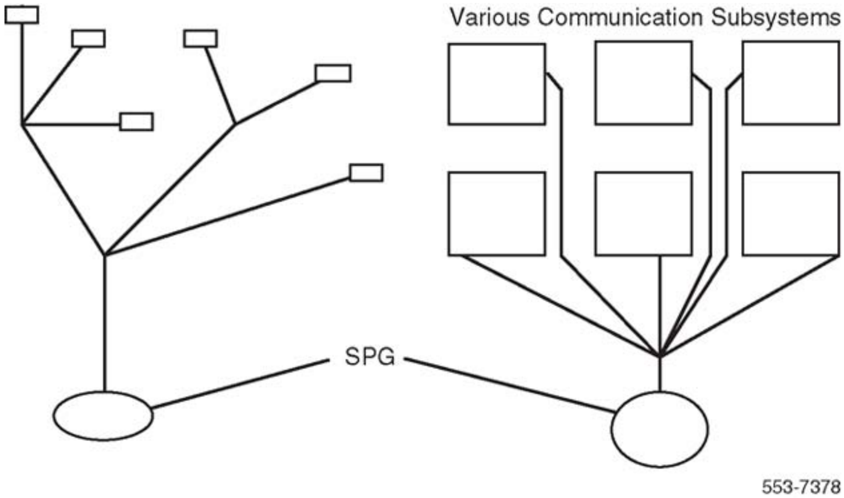

Figure 30: Single point grounding

SPG requirements

SPG requirements are divided into the following major categories:

- Safety on page 93
- Protection on page 94
- EMC on page 94
- Installation and maintenance considerations on page 94
- Powering on page 94

Safety

To ensure a safe working environment for trained company personnel, the customer premises grounding system must be able to dissipate surge energies (such as lighting strikes on the outside plant). In addition, the grounding system must be designed to ensure that fuses or breakers operate to disrupt any excessive current flow caused by a power fault.

Protection

A proper ground is essential for system protection equipment. This includes grounding for outside plant cable shields and protectors, as well as the grounds associated with framework, battery, and logic references.

EMC

Grounding must be considered at all times to ensure good Electromagnetic Compatibility (EMC), emission and susceptibility performance.

Installation and maintenance considerations

If included as part of the initial electrical installation for the customer premises, a grounding system is cost effective to install and maintain. Adding a grounding system after the initial installation is complete can be difficult and costly.

Powering

When planning the grounding system, consider the powering options for the equipment. Look at whether the equipment is backed up with batteries or a UPS. The grounding and powering of all equipment associated with the telecommunications system should be considered as one large system.

Types of grounding

The system has several different grounds and signal returns that are generally referred to as grounds. The types of grounds include:

- Safety (personal hazard) ground on page 95 (see Safety (personal hazard) ground on page 95)
- Logic return on page 98 (see Logic return on page 98)
- Battery return (for DC systems)

Figure 31: AC power multiple-column distribution ACEG on page 97 and Figure 32: AC power multiple-row distribution ACEG on page 98 illustrate examples of power and ground connections in several AC system configurations.

Safety (personal hazard) ground

If conduit is used to connect AC power from a service panel to the pedestal, it must contain an insulated ground wire (green) that is #6 AWG or larger size. If a cord-and-plug connection is used, a separate safety ground must be provided.

The safety ground is required to reduce the risk of electric shock to personnel and avoid system malfunctions under the following conditions:

- contacts between telephone wire and AC current elsewhere in the building while the AC input cord is unplugged
- lightning transients when the cord is unplugged
- stray grounds during normal operation

The safety ground, also known as frame ground or chassis ground, must be an insulated wire #6 AWG or larger, and must connect to both the pedestal safety ground lugs and the service panel ground bus. In all systems, one 30 A circuit is required for each column. Isolation, as required by NEC 250-74 and 384-27 (exception 1), is preferred.

An SPG is an isolated ground (IG) bus or AC equipment ground (ACEG) bus in the service panel or transformer. It may also be a separate external bus bar that connects at a single point to the service panel or transformer. Figure 31: AC power multiple-column distribution ACEG on page 97 and Figure 32: AC power multiple-row distribution ACEG on page 98 show an isolated ACEG as the SPG.

Depending on the distance between columns (and cabinets in upgraded systems) and the service panel, the safety ground wiring may be daisy-chained or run independently from each column (or each row) to the ACEG. Figure 31: AC power multiple-column distribution ACEG on page 97 and Figure 32: AC power multiple-row distribution ACEG on page 98 show safety ground wiring in daisy-chain configurations.

To implement the SPG, follow these guidelines:

1. All ground conductors must comply with local electrical codes and be terminated in a manner that is permanent, resulting in low impedance connections.
2. All terminations should be readily accessible for inspection and maintenance.
3. A grounding conductor must be continuous, with no splices or junctions.
4. The insulated grounding wire size must comply with the National Electric Code (NEC) Sections 250-94, 250-95, and 310-15.
5. Conductors must be insulated against contact with foreign (nonAC) grounds.
6. Grounding conductors must be no-load type and carry no current under normal operating conditions.
7. The use of building steel as an integral part of the ground system is not recommended.

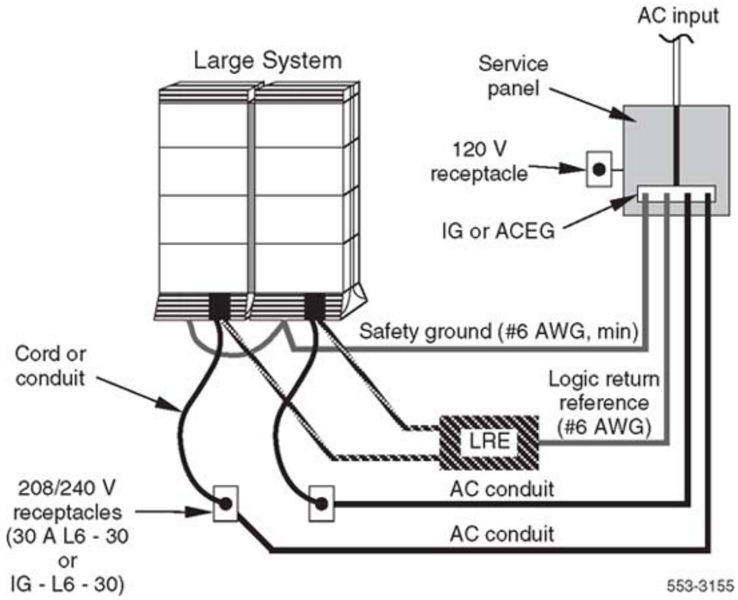

Figure 31: AC power multiple-column distribution ACEG

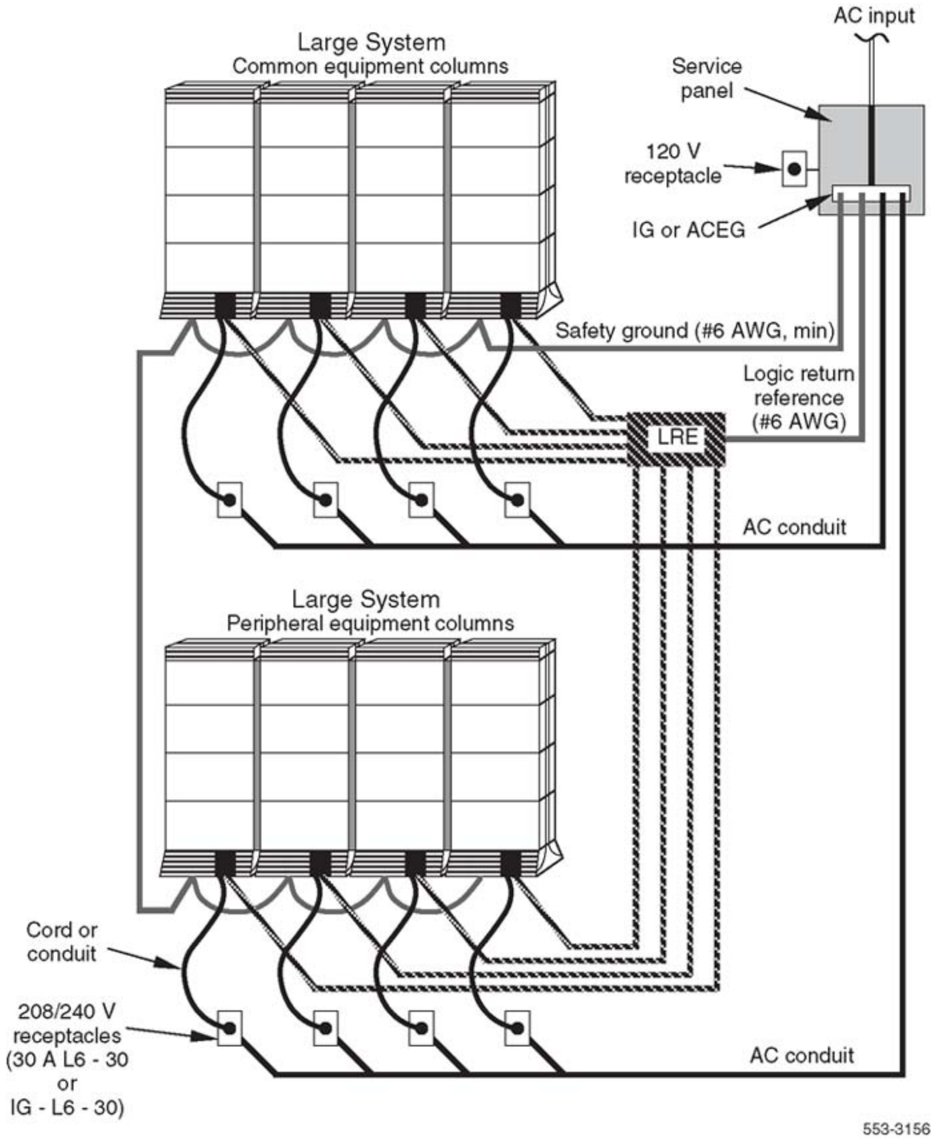

Figure 32: AC power multiple-row distribution ACEG

Logic return

A logic return equalizer (LRE) is a separate bus bar (such as an NT6D5303 or NT6D5304) used to join logic return wires at a common point. A #6 AWG conductor connects the LRE to the ACEG in the service panel. With multiple columns, the LRE is typically located in a nearby rack, in an overhead trough, or under a raised floor. The LRE must be insulated from the AC-grounded support structure. Figure 31: AC power multiple-column distribution ACEG on page 97 and Figure 32: AC power multiple-row distribution ACEG on page 98 show the use of an LRE in multicolumn configurations.

The LRE is a consolidation point for all the logic return grounds. It is connected to the ACEG, which is located within the system's dedicated AC panel. The isolated ground bus within the dedicated AC service panel serves as the "system" SPG. The dedicated AC service panel should be supplied from the building's principal ground source, usually a transformer located within the building. It is at this point that the neutral-to-ground bond is performed. The live, neutral, and grounding conductors are supplied, all together within a single conduit, from the building's principal ground source to the dedicated AC service panel. The dedicated AC panel should service all the communication equipment and any logically interconnected devices (such as modems, TTYs, multiplexers, etc.). This ensures that all equipment has the same ground reference.

Identifying good grounds

The main ground consists of rods or plates. It is considered a good ground when the resistance of the rods or plates to ground is as low as practical. A recognized industrial standard is 4 ohms.

Usually a visual inspection will suffice to ensure that the connection of the ground conductor to the main ground is soundly made. It is possible to verify the quality of the ground by using a three-terminal tester. Figure 33: Three-terminal testing on page 99 illustrates this procedure.

Refer to the three-terminal tester manufacturer's handbook for testing instructions.

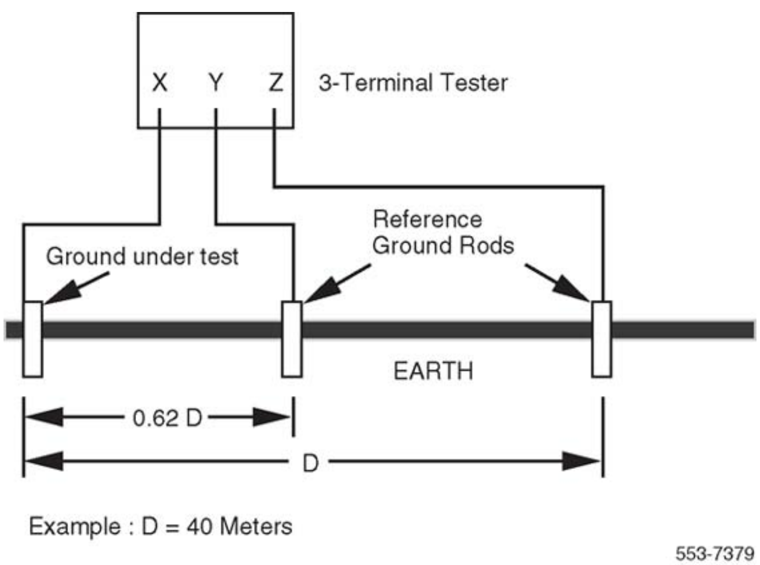

Figure 33: Three-terminal testing

Circuit protection

RS-232 port protection

RS-232 type interfaces are susceptible to induced lightning damage when hardwired lines are run building to building. As little as 25 V can cause damage. Typically only pins 2 (send), 3 (receive), and 7 (signal ground) are connected end-to-end via twisted, shielded pairs.

Although the RS-232 specification supports only 15 m (50 ft) of operation, many applications successfully pass data at much longer distances. However, problems arise when different grounds are used at the two ends of the cable. Grounding at both ends will cause a ground loop current to flow in the shield due to the fact that each ground point will most likely be at a different potential. This current flow will induce a voltage onto the signal or data lines, resulting in erroneous data or fault conditions.

To prevent the creation of a current loop, the shield must only be grounded at one end. In general, this grounding takes place at the system end. SDI ports must be connected to the I/O panel at the rear of the switch. RS-232 cables should then be connected to the I/O panel. RS-232 cables should never be connected directly to the connector on the SDI pack.

A modem or isolator must be installed for all RS-232 devices not connected to the ACEG.

Off-premises line protection

All voice and data lines that run externally from the building containing the system must have proper line protection. The cable sheath must be connected to the SPG.

Power service panel

Power service panels must meet the following requirements or be modified when used for the system:

1. Panels should be located in the equipment room.
2. No lights, air conditioners, heaters, generators, or motors may be connected to this service panel.

Power systems

Large Systems feature a modular power distribution architecture. As part of the modular design, the power system provides:

- a pedestal-mounted power distribution unit providing input voltage (AC or DC) to each module and protection from current overload
- power supplies in each module
- a universal quick-connect power wiring harness that carries the power and monitor signals to the power supplies in each module
- modular backup capabilities on a per column basis

The terms "AC system" and "DC system" refer to the type of power brought into the pedestal and distributed within the system to the module power supplies. Figure 34: AC-powered system on page 102, Figure 35: AC-powered system with reserve power on page 103, and Figure 37: DC-powered system on page 109 starting on Figure 34: AC-powered system on page 102 show the basic power distribution for AC and DC systems. All system options are available in both AC power and DC power versions.

To understand the system power architecture, consider the distinction between the "internal" and "external" power components.

Internal power components

Internal power components are contained within the system itself. They form an integral part of the power subsystem. They include the power distribution unit (PDU) in the pedestal, the power wiring harness, and the module power supplies.

Although the PDU and module power supplies differ in AC- and DC-powered systems, power distribution is similar: power is input to the pedestal and distributed to individual power supplies in each module. In AC-powered systems, the module power supplies convert the AC voltage to several usable DC voltages; in DC-powered systems, the module power supplies convert the DC voltage to several usable DC voltages. Except for the power components and the power wiring harness, all other functional elements within the system (such as card cages, backplanes, circuit cards, and system monitor) are identical in both AC and DC systems.

External power components

External power components are outside the system columns. If reserve power is not required, AC-powered systems have no external components; AC systems plug directly into the commercial AC power source. If an Uninterruptible Power Supply (UPS) is installed for reserve power, it is considered an external power component. All DC systems are powered by rectifiers that are external to the system.

AC power

AC-powered systems require no external power components and can plug directly into the commercial utility power source (see Figure 34: AC-powered system on page 102). AC powering requires a single conversion from the AC input voltage to the DC voltages required by the system. Power supplies in each module perform this conversion.

AC-powered systems are well-suited for applications that do not require reserve power. They are also recommended for small to medium-sized (two columns or less) systems that require reserve power with backup times ranging from 15 minutes to 8 hours.

If reserve power is required with an AC-powered system, a UPS is installed in series with the AC power source (see Figure 35: AC-powered system with reserve power on page 103). AC-powered systems that do not require long-term backup can benefit from a UPS with short-term backup. A UPS can provide power conditioning during normal operation, including protection against sags, brownouts, and other low-voltage transient conditions that cause most power disturbances.

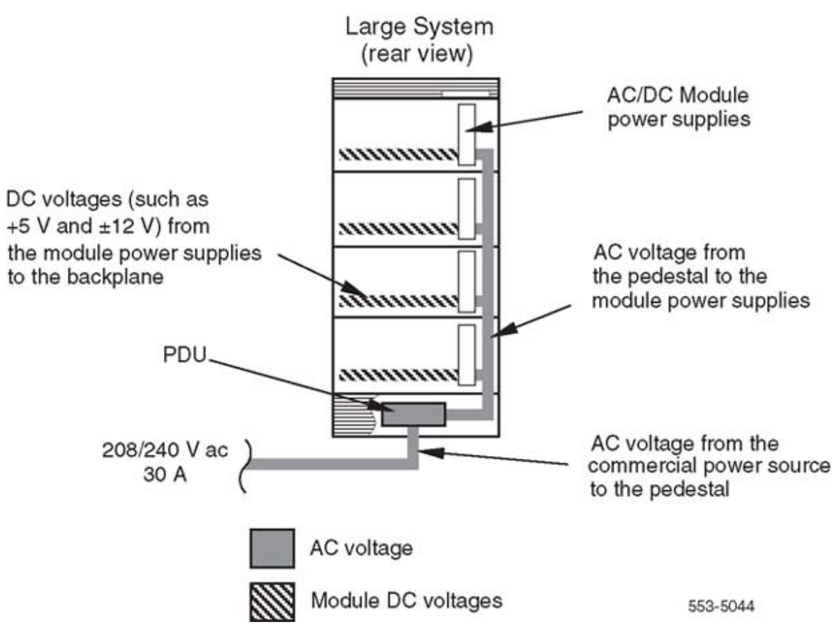

Figure 34: AC-powered system

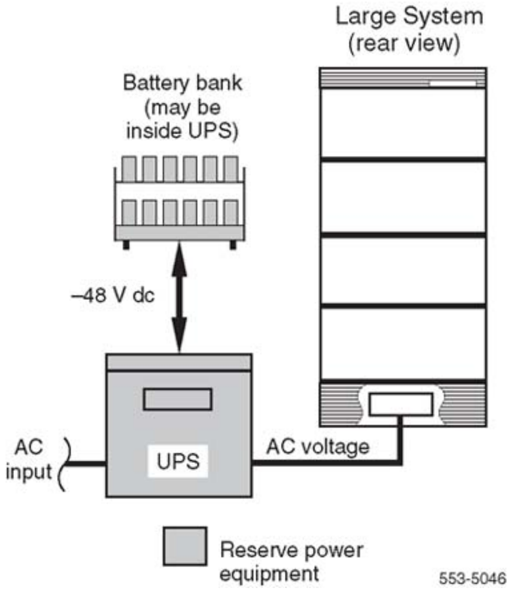

Figure 35: AC-powered system with reserve power

AC power input specifications

AC power supplies operate from a nominal input of 208 to 240 volts AC, single-phase. While the actual input range of the AC power supplies is 180–280 V, no restrapping the power supplies is required if the input line voltage is within 208–240 V.

AC-powered systems require one IG-L6-30 or L6-30 receptacle for each column within 2.4 m (8 ft) of the column's pedestal. Each column comes equipped with one 30 A cord and plug.

Do not use ground fault circuit interrupt (GFCI) devices on AC power circuits.

As an alternative to using the power cord and plug, AC input to the PDU may be wired directly. Use #10 AWG conductors routed through 1.9-cm (3/4-in.) conduit. Connect the conductors to the input terminals on the field wiring terminal block in the PDU for a 240 V AC input, as indicated in Table 5: AC input connections on page 103.

Table 5: AC input connections

AC input conductor	PDU terminal
Hot – Phase I	L1
Hot – Phase II	L2
Safety Ground	GND

All AC input power wiring must contain a separate safety ground conductor (green wire). Avaya strongly recommends a dedicated AC supply that runs uninterrupted from the building primary source to a dedicated equipment room service panel.

Follow all applicable electrical codes if the AC input is wired directly to the PDU.

If reserve power is used, install the UPS, along with its associated batteries (which may be internal or external to the unit), in series with the commercial power source. The system then plugs into the UPS (see Figure 35: AC-powered system with reserve power on page 103). Consult the UPS manufacturer for requirements of the UPS power input receptacle.

AC internal power distribution

Figure 36: AC internal power distribution (rear of column) on page 105 shows the elements of the AC internal power system. Components of the distribution system include the power distribution unit (PDU), module-to-module power harness, module power distribution unit (MPDU), MPDU-to-backplane power harness, and module power supply.

Input power wiring connects to the field wiring terminal block in the back of the PDU. The output power harness connects the field terminal block to the first module. The module-to-module harness distributes power to the MPDU in each successive module. The MPDU-to-backplane harness distributes power from the MPDU to the module power supply and ringing generator, if equipped. The module power supply converts the AC voltage to the DC voltages required by the circuit cards in the module.

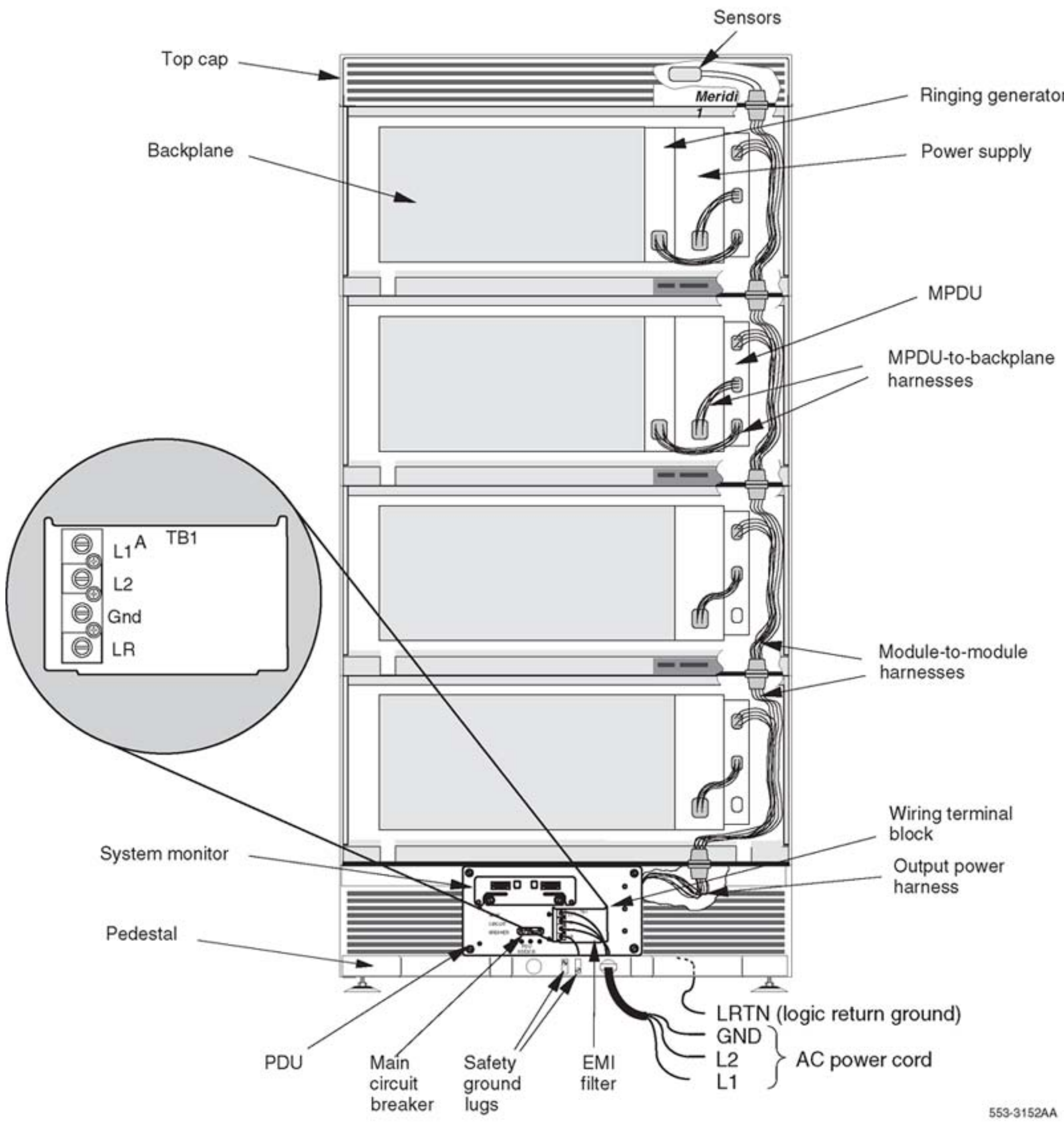

Figure 36: AC internal power distribution (rear of column)

Power distribution unit (PDU)

Located in the rear of the pedestal, the PDU serves several functions, but primarily it serves as a power distribution point for the entire column. The field wiring terminal block provides the connection point for the AC input wiring. The electromagnetic interference (EMI) filter (required for regulatory compliance) keeps EMI from radiating outside the confines of the column. The circuit breaker, which is the main circuit breaker for all modules in the column, protects the column if there is a current or thermal overload. The internal terminal block provides a distribution point for power output wiring to the modules. The power/signal harness (not shown in Figure 36: AC internal power distribution (rear of column) on page 105) provides power and signal interconnections for the blower unit and system monitor. The system monitor power supply provides +5 V power to the system monitor even when the main circuit breaker is off. The output power harness relays power from the pedestal to the module(s) above it.

The system monitor is housed in the PDU. The system monitor is powered by a small AC power supply, which is not connected to the circuit breaker for the column.

Intermodule harnesses

Several power harnesses conduct the input voltage throughout the column (see Figure 36: AC internal power distribution (rear of column) on page 105). The module-to-module harness connects to the MPDU in each module and to the module above. The MPDU-to-backplane harness distributes power from the MPDU to the module power supply and ringing generator, if equipped, through backplane power connectors.

Module power distribution unit (MPDU)

An MPDU provides the circuit breakers that provide current protection at the module level, so only a faulty module is shut down while others remain functional. Table 6: MPDU, power supply, and module compatibility on page 106 lists the MPDUs, power supplies, and compatible modules.

Table 6: MPDU, power supply, and module compatibility

MPDU	Power supply	Module
NT8D56AA	NT8D29	NT4N41BB Core/Network
NT8D57AA	NT8D06 PE NT8D21 (ring generator)	NT8D37 IPE

Module power supplies

In each module, input voltage is carried through the backplane harness to the module power supply, where it is converted to the voltages required by the circuit cards in the module. Table 6: MPDU, power supply, and module compatibility on page 106 lists the compatibility between module, MPDUs, and each power supply.

Table 7: Output voltages and currents for AC power supplies on page 107 lists the output voltages and currents for AC module power supplies.

Table 7: Output voltages and currents for AC power supplies

Module	Output volts (V AC)	Output amperes (A)	Output volts/ volt-amperes (V AC/VA) (see Note)	Output frequency (Hz)
NT8D06 PE Power Supply	+5.1 +8.5 +10.0 –10.0 +15.0 –15.0 –48.0	28.00 4.00 0.50 0.50 17.00 15.00 7.70	—	—
NT8D21 Ringing Generator	–150.0 +70.0 +80.0 +86.0	0.20 0.127 0.111 0.103	70/8 80/8 86/8	25/50 25/50 20/25
NT8D29 Power Supply	+5.1 +3.3 +12.0 –12.0	60.00 5.00 2.50 1.00	—	—
Volt-amperes (VA) is for the ringing power.				

DC power

DC power systems deliver DC to the pedestal of the system. AC-powered systems accept AC at the pedestal and distribute AC to the power supplies located in each module.

DC-powered systems require an external DC power plant consisting of rectifiers (also called chargers or AC/DC converters) and power distribution and control equipment. The external rectifiers connect directly to a commercial AC power source (see Figure 35: AC-powered system with reserve power on page 103). DC-powered systems require a double conversion: the rectifiers convert the AC voltage to –48 V DC, which is distributed by the PDU in the pedestal to the power supplies in the modules. The power supplies convert –48 V DC to other DC voltages required in each module.

Batteries are generally used with DC-powered systems because the traditional method for powering telecommunications uses rectifiers to continuously charge a bank of batteries, while the system power "floats" in parallel with the battery voltage. However, batteries are only used if reserve power is needed. Figure 38: DC-powered system with reserve power on page 109 shows a DC Rectifer system with reserve power equipment.

Complete systems — including DC power plants — can be provided by Avaya. Systems can also be configured to connect to an existing power plant provided by the customer.

Consider DC systems for the following:

- most Large Systems (any system larger than two columns)
- most systems with long-term reserve power requirements (usually eight hours or more)
- when the customer site has an existing DC power plant or batteries available

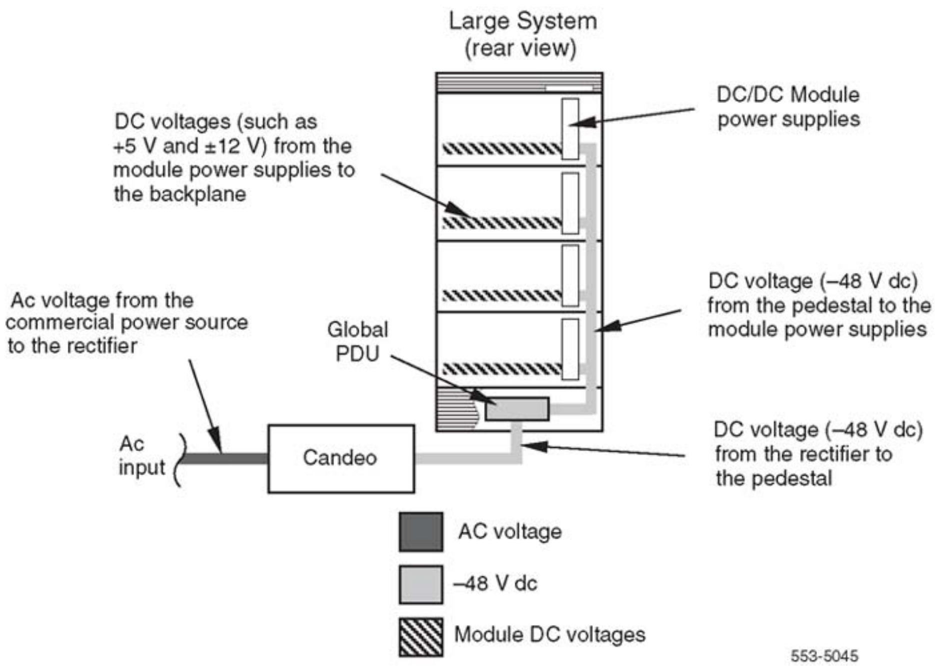

Figure 37: DC-powered system

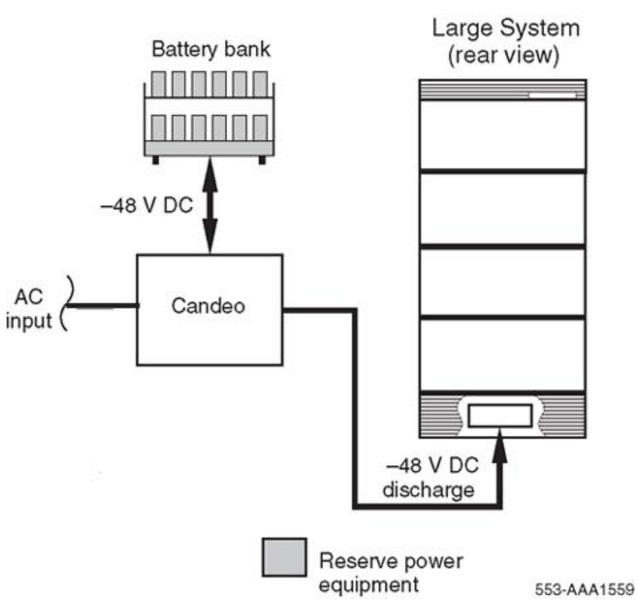

Figure 38: DC-powered system with reserve power

The DC power system must be able to provide the required current and operate within the specifications listed in Table 8: Input specifications DC power system on page 110. For additional battery voltage requirements, see Table 12: Battery requirements on page 118.

Table 8: Input specifications DC power system

Input	Pedestal	Battery
Maximum range	–42 to –56.5 V	–42 to –56.5 V
Expected nominal (24 stationary cells)	—	–52.08 V
Expected nominal (23 sealed cells)	—	–51.75 V
Expected nominal (24 sealed cells)	—	–54.00 V
Noise (max C msg)	—	22 dBrnC (See Note)
Without battery, C msg (max) is 32 dBrnC.		

Input power specifications

DC power plants require one separate AC input per rectifier, within 1.8 m (6 ft) of the rectifier. The number and type of rectifiers used determine the total requirements for commercial AC power input.

Do not confuse the output rating of the rectifiers in DC amps with input requirements in AC amps.

Internal power distribution

Power distribution in the DC-powered system (see Figure 39: DC internal power distribution (rear of column) on page 111) consists of the NT4N49AA PDU, the module-to-module power harness, the module-to-backplane power harness, and the module power supply.

DC power cables connect to the field wiring terminal block, where an output power harness carries the input voltage to the first module. Module- to-module harnesses distribute DC voltage to successive modules. Module-to-backplane harnesses distribute DC voltage to the module power supply and ringing generator, if equipped. In each module, the module power supply converts the DC input voltage to the several DC voltages required by the circuit cards in the module.

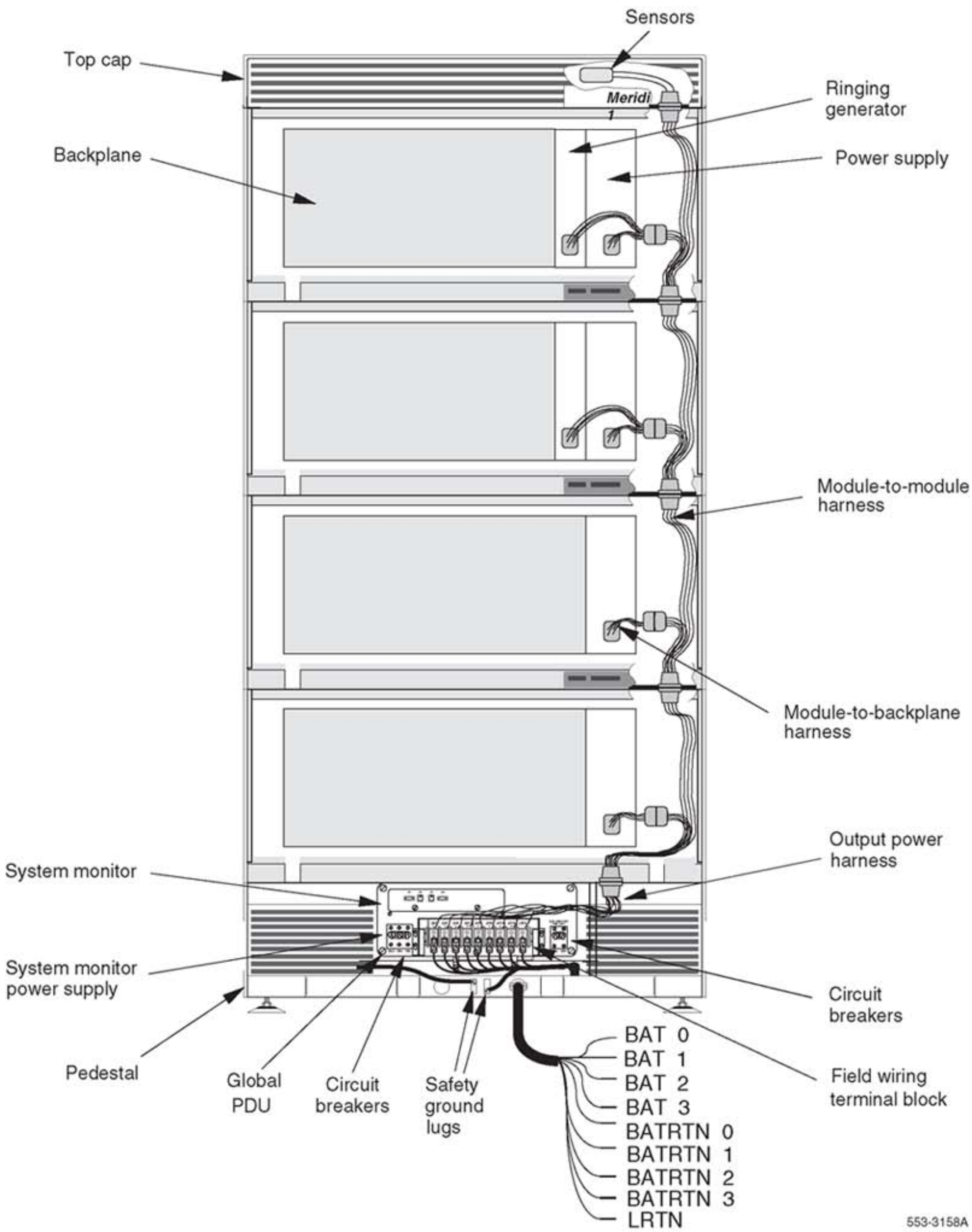

Figure 39: DC internal power distribution (rear of column)

Power distribution unit

TheNT4N49AA PDU, located in the rear of each pedestal, distributes power to the entire column. The PDU provides a common distribution point for the input voltage. The output power harness connects the pedestal to the first module. Individual wiring harnesses carry the current to each successive module. The power/signal harness (not shown in Figure 39: DC internal power distribution (rear of column) on page 111) provides power and signal connections between the blower unit and system monitor.

In the event of a current overload, one of four circuit breakers located in the PDU protects each module. A fifth circuit breaker provides protection for the whole column in the event of a thermal overload. The system monitor power supply provides +5 V power to the system monitor (even when the PDU circuit breakers are off).

See NT4N49AA 4-Feed PDU on page 120 for additional information.

Intermodule harnesses

Several power harnesses conduct the input voltage throughout the column. The module-to-module harness consists of the module connector, which distributes power to the module or modules above it, and the backplane connector, which distributes power through the module-to-backplane harness to each module power supply.

Module power supplies

In each module, –48 V is received through the module-to-backplane distribution harness and converted by the module power supply to the necessary voltages for the individual module. There is an on/off switch on each power supply for safe operation and easy maintenance.

There are three DC module power supplies:

1. The NT6D40 PE Power Supply provides power to circuit cards and talk battery to lines and trunks.
2. The NT6D41 CE Power Supply provides power to circuit cards.
3. The NT6D42 Ringing Generator provides –150 or –100 V DC message waiting lamp voltages for 500/2500-type telephones. It can provide ringing power to 48 ringers simultaneously.

Table 9: Power supply and module compatibility on page 112 lists power supply compatibility. Table 10: Output voltages and currents for DC power supplies on page 113 lists the output voltage and currents for DC power supplies.

Table 9: Power supply and module compatibility

Power supply	DC Module
NT6D40 PE	NT8D37 IPE

Power supply	DC Module
NT6D41 CE	NT8D35 Network
NT6D42 ring generator	NT8D37 IPE

Table 10: Output voltages and currents for DC power supplies

Power supply	Output volts	Output amperes (A)	Output volts/ volt-amperes (V AC/VA) (Note)	Output frequency (Hz)
NT6D41 CE Power Supply	+5.1 +12.0 –12.0	60.00 3.50 1.00	—	—
NT6D42 Ringing Generator	–100.0 –150.0 70.0 80.0 86.0	0.20 0.20 0.127 0.111 0.103	70/16 75/16 80/16 86/16	20/25/50 20/25/50 20/25/50
Volt-amperes (VA) is for the ringing power.				

External power distribution

A variety of rectifiers and distribution equipment can be used to supply external DC power. Existing customer equipment can be used or a system that Avaya either supplies or recommends, such as the Small or Large DC Rectifer. The Small DC Rectifer is appropriate for Single Group or Multi Group systems that do not require more than 300 A, while the Large DC Rectifer is suitable for larger systems. In all cases, equipment for rectification and distribution is required, while reserve batteries are optional.

For installation procedures, see *Avaya Communication Server 1000M and Meridian 1 Large System Installation and Commissioning, (NN43021-310)*.

Reserve power

Reserve power is available for both AC and DC systems. When selecting reserve power equipment, consider the following:

- future system growth
- the maximum time backup power is required
- existing power system capacity

- the space and thermal environment (air conditioning)
- other equipment, such as lights and alarm systems

Reserve power for AC systems is provided by UPS units, installed in a series with the commercial power source.

DC systems use the traditional telecommunications powering method: external rectifiers (AC/DC converters) continuously charge a bank of batteries while the system power "floats" in parallel on the battery voltage.

AC reserve power

A UPS generally consists of a combination battery charger (AC/DC converter) and inverter (DC/AC converter), along with associated batteries. The batteries may be internal or external to the UPS. A UPS is not a standby power source, but an on-line 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, first calculate the total power requirements of the column (or columns) supported by the UPS, as described in Calculating system power requirements on page 130. Convert the real power in watts (W) to complex or "apparent" power in volt-amperes (VA) by dividing the real power by the typical system power factor of 0.6. Then size the UPS in terms of its rating in VA (or kVA). For AC-powered systems, EC calculates the system power consumption in both watts and volt-amperes.

$$VA = \frac{W}{0.6}$$

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

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

UPS interfacing

A UPS must meet the following requirements in order to be used with a system:

1. The UPS specifications must meet the commercial power specifications of the system:
    a. nominal output voltage range of 208–240 V AC, with a total input range of 180–250 V AC
    b. nominal frequency of 50–60 Hz, with a total range of 47–63 Hz
    c. Total Harmonic Distortion (THD) of 5%, with 3% on any single harmonic, of the AC sine wave
2. The UPS must be able to handle nonlinear loads (the AC module power supplies are a switched-mode design) and have a current crest ratio of 3.0 or greater.
3. The UPS must be UL listed and certified under FCC Part 15, Subpart J as a Class A device.
4. The UPS must have a 30 A, 250 V locking power receptacle (L6-30) for each column to be powered.
5. The UPS must meet ANSI standard C62.41 and IEEE standard 587-1980, class A and B, for transient surge suppression.

It is convenient for the UPS to have one or more 120 V power outlets (5-15R) for auxiliary devices that must have backup power, such as the Power Failure Transfer Unit (PFTU) power supply.

UPS installation

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

UPS installation can be complex. Avaya recommends taking advantage of vendor training programs.

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 allows the system to 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 40: AC reserve power configuration on page 116 shows a general block diagram of a UPS installation and associated wiring.

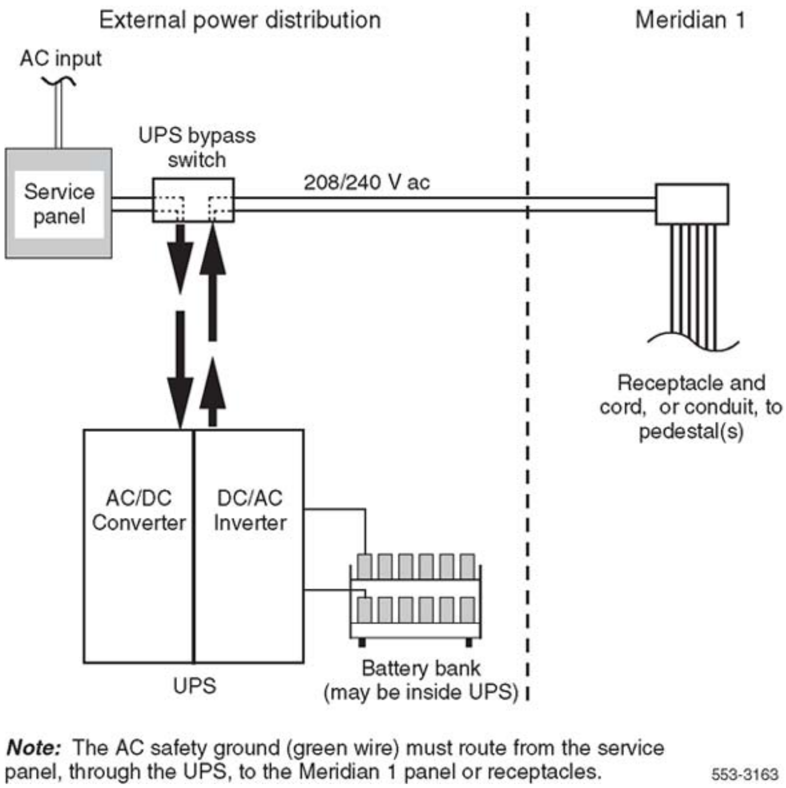

Figure 40: AC reserve power configuration

Power conditioning

The term "power conditioner" refers to a wide variety of power protection or power quality improvement devices, such as low-pass filters, surge arrestors, line voltage regulators, and isolation transformers. Most of these devices can help prevent power line spikes and surges, and some isolation transformers can provide good noise rejection.

Although most power conditioning devices do not provide energy storage for undervoltage conditions, they can help prevent surges and other overvoltage conditions that can cause permanent damage to equipment.

When choosing UPS protection and power conditioning equipment, remember that over 90% of power disturbances in the US are undervoltage conditions such as sags and outages. When there are US power disturbances:

- 87% are power sags; 90% of these last 0.53 seconds or less
- 7.4% are impulses or spikes lasting less than 100 microseconds
- 4.7% are longer-term power failure; 90% of these last 4.2 hours or less, 75% last 40 minutes or less, and 50% last 38 seconds or less
- 0.7% are surges lasting more than 100 microseconds

Low-voltage transients occur most frequently and may cause temporary loss of operation. High-voltage transients occur much less often, but can cause damage to equipment as well as loss of operation.

Carefully consider the type of power line protection needed for the installation under consideration. A power conditioner can help provide overvoltage protection, but a UPS can provide both overvoltage and undervoltage protection, though usually at a higher price.

Alarm monitoring

Avaya offers a UPS-to-system monitor interface cable for each of the product lines that have been tested for system compatibility. The system monitor interface is not supported for other vendors. Table 11: UPS-to-system monitor alarm cables on page 117 lists the UPS-to-system monitor alarm cables that are available. UPS systems are not offered by or available through Avaya, but can be purchased directly from vendors or through authorized distributors.

The alarm interface consists of:

- an "Inverter On" signal to indicate that the commercial power is interrupted and the UPS alone is supplying power to the system
- a "Summary Alarm" signal to indicate a fault or alarm condition at the UPS

Table 11: UPS-to-system monitor alarm cables

UPS vendor	Order Code	Quantity
Alpha Technologies	NT8D46AU	One per UPS
Best Power Technology	NT8D46AJ	One per UPS

UPS vendor	Order Code	Quantity
Exide Electronics	NT8D46AQ	One per UPS

DC reserve power

Reserve power for DC systems is provided by adding batteries to the external power distribution system. Calculate reserve battery capacity as described in UPS sizing on page 114. This determines the total ampere-hour requirements of the batteries. (See also Calculating system power requirements on page 130.)

To comply with safety requirements, read and fully understand the following documents before working with any battery systems:

- OSHA "Material Safety Data Sheet." This must be posted to meet OSHA requirements. This document outlines safe reserve battery handling procedures.
- National Electric Code 645-10. This document outlines requirements for the installation of AC and DC power kill switches to battery systems in certain environments.

Current requirements

The DC current required for battery reserves is based on the total system power requirement. For new installations, you can determine power and battery requirements from data provided by Avaya. For existing installations, see Calculating system power requirements on page 130 for information about calculating current required for battery reserves.

Batteries

The reserve battery capacity required depends on the system line size (load), the time the reserve supply must last in the event of a power failure, and the battery end voltage. See Table 12: Battery requirements on page 118 for reserve battery float voltage and equalization voltage guidelines. These voltages must never be more negative than – 56.5 V.

Table 12: Battery requirements

Battery configuration	Float voltage (V)		Equalize voltage (V)	
	Cell	Bank	Cell	Bank
24 stationary cells	–2.17	–52.08	–2.25	–54.00
23 sealed cells	–2.25	–51.75	–2.35	–54.05

Battery configuration	Float voltage (V)		Equalize voltage (V)	
	Cell	Bank	Cell	Bank
24 sealed cells	–2.25	–54.00	–2.35	–56.40

Lead-calcium/absolyte batteries

Battery package provisioning is based on the number of ampere-hours required. Since battery package ampere-hour ratings are generally given at an 8-hour discharge rate, adjustment factors are required to determine the required battery package. Table 13: Adjustment factors for lead-calcium and absolyte batteries on page 119 lists adjustment factors for lead-calcium and absolyte batteries. These factors are based on the discharge rates of the respective battery types from a specific supplier. Discharge characteristics may vary by manufacturer.

Table 13: Adjustment factors for lead-calcium and absolyte batteries

Reserve hours	Lead-calcium factor	(Sealed) Absolyte factor
1	3.0	1.8
2	4.0	3.1
3	5.0	4.2
4	5.9	5.2
If a system requires more than 10 hours of backup, the factor is linear. For example, if 15 hours are required, the factor is 15.		
5	6.9	6.2
6	7.7	7.1
7	8.5	7.8
8	9.3	8.5
9	10.1	9.4
10	10.9	10.2

Calculate battery requirement using this formula:

$$Ahr = I_L \times F_{adj}$$

- Ahr = battery requirement in ampere-hours
- I_L = system load, in amps
- F_{adj} = appropriate adjustment factor from Table 13: Adjustment factors for lead-calcium and absolyte batteries on page 119

When using lead-calcium or sealed batteries, calculate battery recharge time using this formula:

$$T = \frac{Ahr \times 1.15}{I_{RO} - I_L}$$

- T = battery recharge time
- Ahr = battery capacity in ampere-hours
- I_L = total system load, in amps
- I_{RO} = total rectifier output, in amps

Other battery considerations are:

- Not all sealed cells require equalization, but equalization voltage can be used for fast charging. Use a battery end voltage of 44 V when choosing battery banks.
- Use these electrical noise limitations for a battery bank:
    - 20 mV rms maximum ripple
    - 32 dBrnC maximum noise
- CEMF cells are not recommended because the noise they generate is unacceptable.

NT4N49AA 4-Feed PDU

The NT4N49AA 4-Feed PDU supports independent power feeds to each of four modules in a stack if required. However, in a typical installation where independent power feeds are not required, two jumper wires are provided to jumper adjacent battery leads. When the jumper wires are used, the four-wire PDU effectively provides the same "shared" power configuration provided by the existing DC PDU. Therefore, the new PDU is backward compatible and can replace an existing PDU unit in a stack if required.

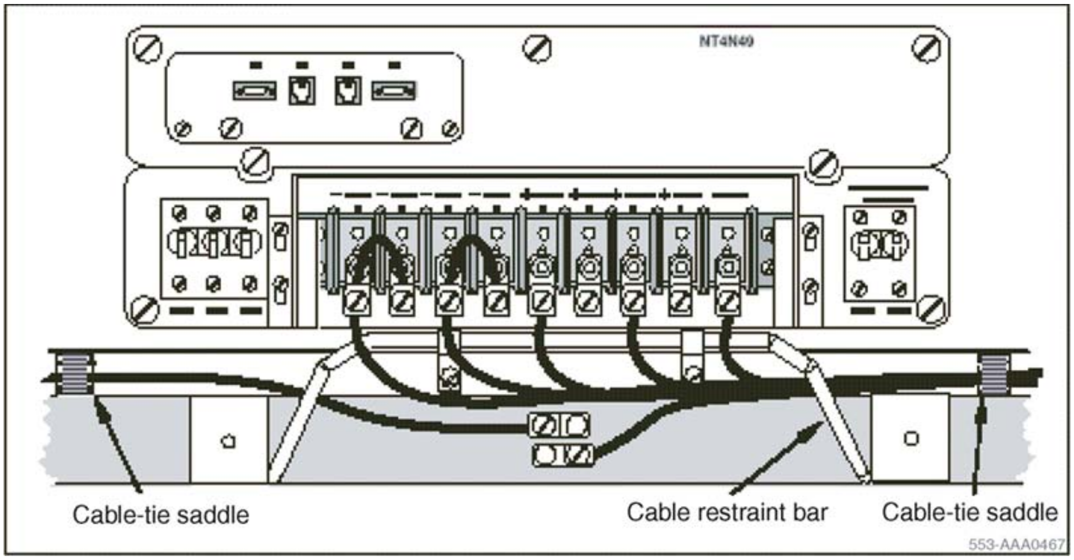

Figure 41: Standard two-feed wiring

The NT4N49AA DC PDU:

- Supports four input circuits, implemented through the following terminal configuration:
    - four (negative) battery leads
    - four return leads
    - logic return lead
- Is fully backward compatible with the existing PDU it is replacing.
- Supports independent power feeds to each of four modules.

The four breakers (one for each module) in the existing DC PDU (NT4N50AA) are rated at 18 A each. The same breakers in the 4-feed PDU are rated at 28 A.

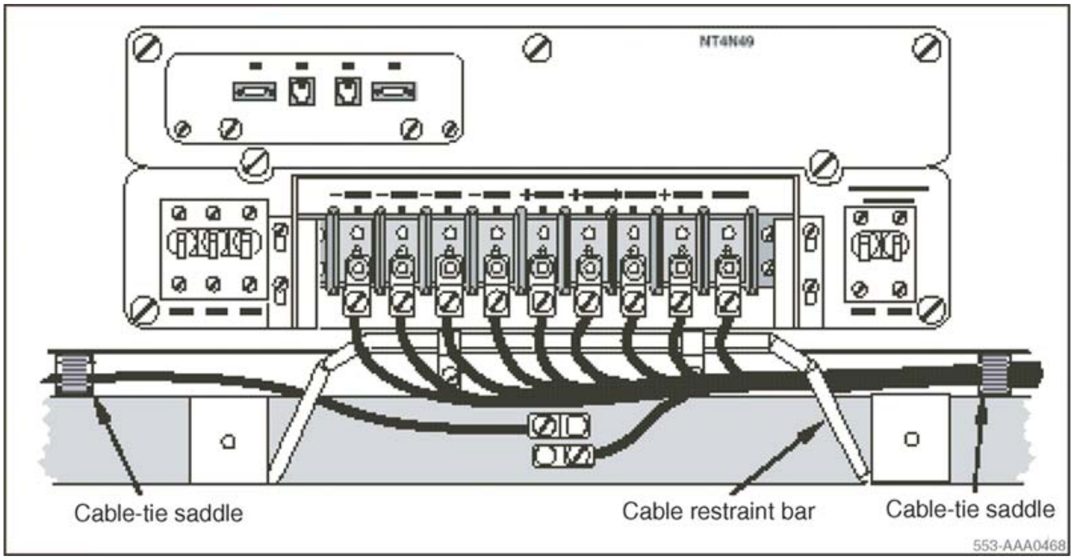

Figure 42: Optional 4-feed wiring

PDU Connections

A readily accessible disconnect device for input power is required.

Caution:

Damage to Equipment

DC power for the pedestal must be provided with circuit protection of 30 A for each feed (– BAT 0, – BAT 1, – BAT 2 and – BAT 3 (see Figure 43: PDU circuit protection on page 123).

Circuit breakers must be located next to each other and labeled to show that both must be shut off to remove all power to the system.

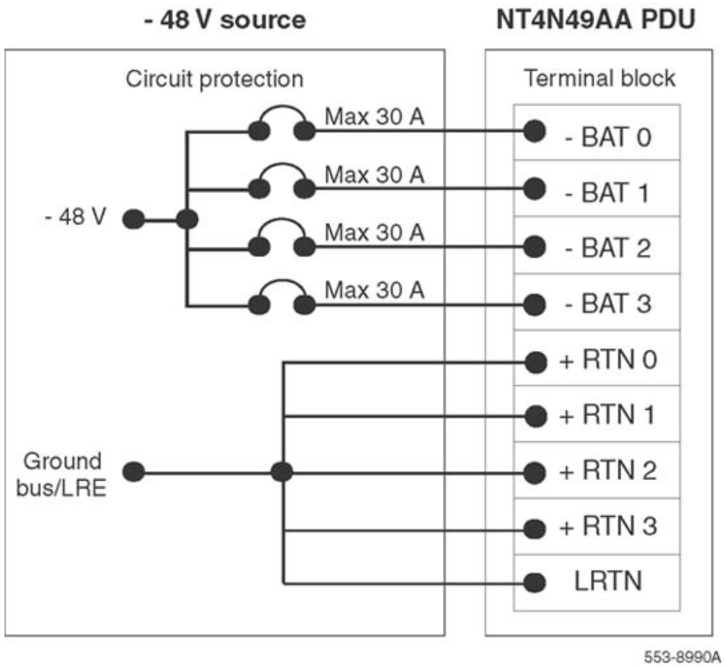

Figure 43: PDU circuit protection

A maximum loop drop of two volts is allowed between the pedestal, or junction box, and the external power equipment. See Table 14: Wire gauge requirements with two 30 A feeds (five wires) on page 123 for allowable wire sizes. See Selecting proper wire size on page 133 for detailed information about calculating wire size.

Table 14: Wire gauge requirements with two 30 A feeds (five wires)

Length	#8 AWG	#6 AWG	Single #4 AWG	Double #4 AWG
0–3 m (10 ft)	Yes	Yes	Yes	Yes
3–6 m (20 ft)	Yes	Yes	Yes	Yes
6–9 m (30 ft)	Yes	Yes	Yes	Yes
9–12 m (40 ft)	Yes	Yes	Yes	Yes
12–15 m (50 ft)	Yes	Yes	Yes	Yes
15–18 m (60 ft)	No	Yes	Yes	Yes
18–21 m (70 ft)	No	Yes	Yes	Yes
21–24 m (80 ft)	No	Yes	Yes	Yes
24–27 m (90 ft)	No	No	Yes	Yes
27–30 m (100 ft)	No	No	Yes	Yes
30–60 m (200 ft)	No	No	No	Yes
over 60 m (200 ft)	No	No	No	No

Length	#8 AWG	#6 AWG	Single #4 AWG	Double #4 AWG
Two 30 A feeds are typically adequate for a column with four modules (five wires total — two 30 A feed pairs plus logic return). If dual conduit is used, the wires must be run in battery/battery return pairs, with one pair in one conduit and the other pair, plus logic return, in the other conduit. Legend: Yes: Wire size is adequate for the distance. No: Wire size has too high a voltage drop and is inadequate for the distance.				

The following equipment is located in the rear of each pedestal in Large Systemcolumns (see Figure 44: DC power equipment in the rear of the pedestal NT4N49AA PDU on page 124):

1. The PDU distributes power to the entire column.
2. The field wiring terminal provides the connection point for wiring brought into the pedestal.
3. A circuit breaker is provided for each module in the column and for the blower unit.

All column circuit breakers will trip if a column thermal overload is detected or a DC power low-voltage condition is sensed. The system monitor checks the column temperature, cooling system status, and system voltage status, and controls alarms and line transfer states accordingly.

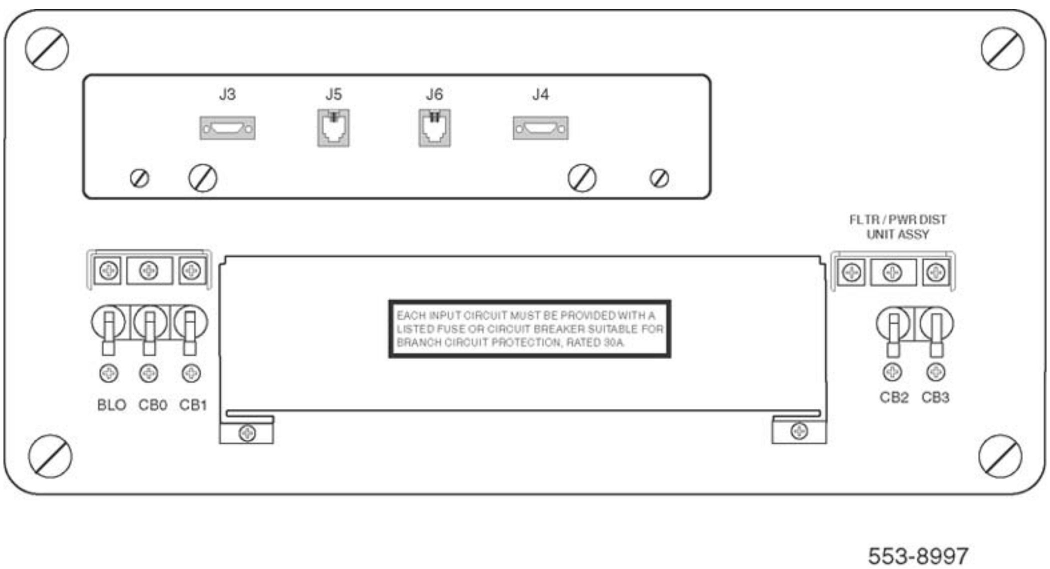

Figure 44: DC power equipment in the rear of the pedestal NT4N49AA PDU

With the NT4N49AA PDU, the safety ground/protective earth wires and all wiring to the terminal block in the PDU must be neatly routed within the cable-tie saddles and under the cable restraint bar at the base of the pedestal (see Figure 45: Cable routing in the rear of the pedestal NT4N49AA PDU on page 126). This ensures that there is room to install the PDU cover, safety cover, and rear grill.

Conduit is not required with the NT4N49AA PDU. However, 1-1/4 or 3/4 in. conduit can be used if local codes or individual installations require it. Conduit can be routed down through the column from overhead racks or up through the floor. Conduit clamps and the hardware to fasten the conduit are provided in the pedestal. If the NT7D0902 Rear Mount Conduit Kit is used, conduit can enter from the rear of the column (above the floor).

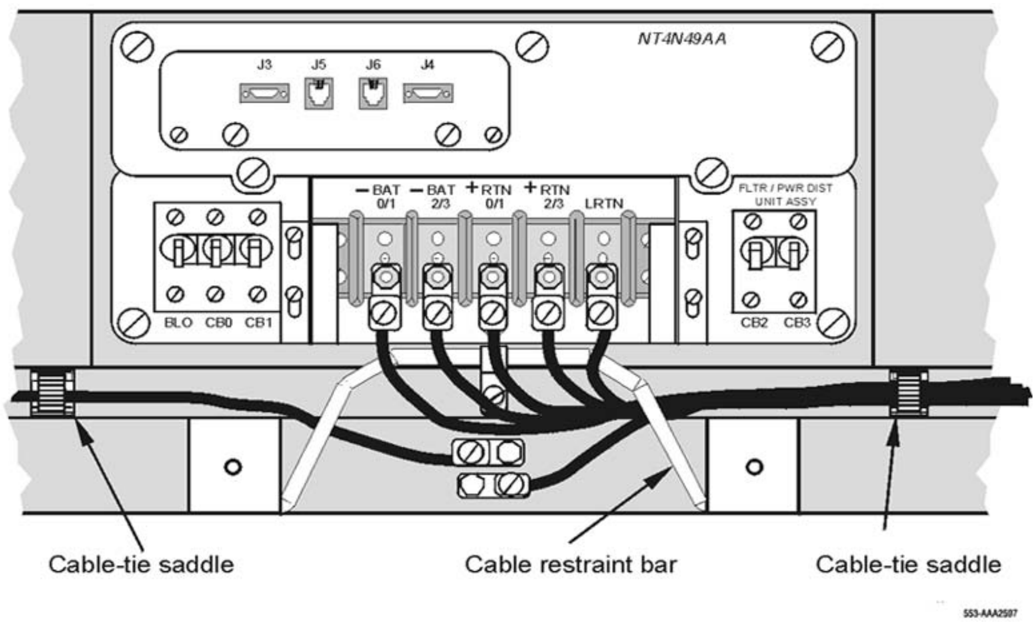

Figure 45: Cable routing in the rear of the pedestal NT4N49AA PDU

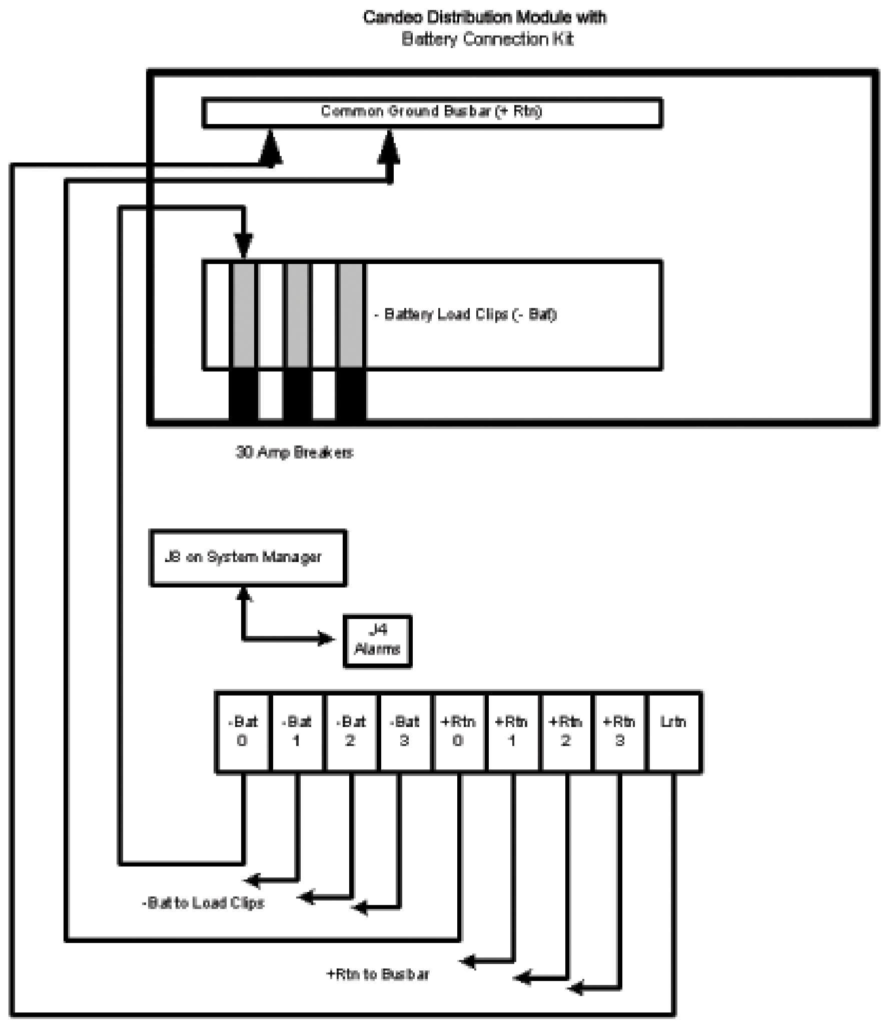

Figure 46: PDU to DC Rectifier connections

Power consumption

Before you can calculate the total power requirements for your system configuration, you must have consumption figures for each component within the system:

- IPE cards -- see Table 15: Power consumption IPE cards on page 128
- Modules -- see Table 16: Module power consumption on page 129

Electrical load varies with traffic load. The following assumptions have been made for the power consumption figures in Table 15: Power consumption IPE cards on page 128 and Table 16: Module power consumption on page 129:

- 50% of digital and analog lines active (18 CCS)
- 75% of trunks active (30 CCS)

The power consumption of digital line cards does not vary greatly with traffic, as it may with analog line cards.

These figures also take into account the average efficiency of the module power supplies.

In general, however, the power consumption figures specified in Table 15: Power consumption IPE cards on page 128 and Table 16: Module power consumption on page 129 are maximum ratings under worst case conditions. Using these figures may result in over-engineering the requirements for UPS. Take power measurements in order to more accurately assess the UPS requirement.

Table 15: Power consumption IPE cards

Circuit card	Power consumption (watts)
NT5K02 Flexible Analog Line card	26
NT5K17 Direct Dialing Inward card	29
NT5K19 DC5/AC15/RAN/Paging Trunk card	29
NT1P62 Fiber Controller card	26
NT7R52 Remote Carrier Interface	26
NT8D01 Controller card-4	26
NT8D01 Controller Card-4 (SMT)	26
NT8D01 Controller Card-2	26
NT8D02 Digital Line card	25
NT8D03 Analog Line card	26
NT8D09 Analog Message Waiting Line card	26

Circuit card	Power consumption (watts)
NT8D14 Universal Trunk card	28
NT8D15 E&M Trunk card	29
NT8D16 Digitone Receiver card	6
NTDU40/41 Media Card	30
NTDW65 Media Card 32S	9
NTDW66 CP PM Signaling Server	30
NT1R20 Off-Premise Station Line card	YTD

Table 16: Module power consumption on page 129 shows power consumption data for each fully configured module. Use this data to calculate rectifier and reserve power (battery).

Table 16: Module power consumption

Module	Power consumption (watts)
NT4N41 Core/Net Module	53.5
NT8D35 Network Module	240
NT8D37 IPE Module	460
Pedestal (with blower unit)	50

Power requirements for IP Phones

IP Phones 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 the BayStack 460. For more information about power over the LAN, see *Avaya Converging the Data Network with VoIP Fundamentals, (NN43001-260)*.

Heat dissipation

Large Systems are equipped with cooling systems and do not have heat dissipation problems under normal applications.

To calculate cooling requirements, consider only power dissipation from the modules.

Btu (thermal load) = total power dissipation × 3.41 For air conditioning purposes, 1 ton = 1200 Btu

Calculating system power requirements

Add the power consumption (in watts) of all equipped modules in order to calculate system power.

The method for calculating system power is based on the number of modules and columns in the system, regardless of how many cards are initially equipped. The method ensures that the external power supply provides adequate capacity, under all conditions and all possible growth scenarios, for the modules installed.

Using a system power consumption worksheet (Worksheet 9:System power consumption on page 391), enter the quantity of each type of module, multiply by the power consumption per module, and then sum the individual module totals to obtain the total real power consumed by the system.

To calculate the current drain, divide the total real power consumption by the nominal input voltage. This gives the system current drain, or load in amperes. The worksheet shows nominal voltages of 208 (AC) input and 52 (DC) output. To calculate current drain for voltages other than those given in the worksheet, divide the total real power consumption by the desired voltage (such as 240 V AC or 54 V DC).

To calculate complex or apparent power (such as for AC wire and panel size or the UPS rating for AC reserve power), divide the total real power in watts by the efficiency (typically 0.6) to obtain the complex power in volt-amperes (see Worksheet 9:System power consumption on page 391).

Power requirements for upgrades

If you are upgrading an installed system, you can determine the total power consumption of the installed system in several ways. Two methods are listed below (the first method is more accurate than the second):

1. Measure current drain for the complete installation under actual operating conditions over at least a two-week period. Determine peak current drain from these measurements.
2. Measure idle (or near idle) current drain for the complete installation. Estimate peak current drain by multiplying the number of idle amperes by 1.5.

When you add or upgrade equipment, use either of these methods to determine existing current drain/power consumption. Use the guidelines in this document to determine the added power consumption.

The existing power plant may have to be replaced or its capacity may have to be increased to accommodate added equipment. Be sure to provide sufficient capacity to accommodate future growth.

System upgrades

Both AC- and DC-powered system upgrade packages are available, although most of the module-level upgrades will be DC.

- Consider an AC upgrade if the existing system is not using reserve power. If reserve power is later desired, one or more Uninterruptible Power Supply (UPS) units can be added.
- If the existing system already has battery backup, or if there is an existing DC power plant or excess rectifier capacity, a DC upgrade package is usually chosen. For DC upgrades, there are several approaches to system powering:
  - An existing external DC power plant may be used as is, or expanded if necessary, to power both the existing equipment and the new equipment.
  - A new external DC power plant, such as a DC Rectifier, may be purchased and installed to power both existing and new equipment.

  For all DC upgrades, carefully measure or calculate the system load of all equipment to make sure that the chosen power system will have enough capacity.
- Consider each upgrade case individually, taking into account the existing equipment, space available at the site, and customer preferences.

Powering upgraded systems from existing rectifiers

Communication Server 1000 upgrades are available in module configurations only. The addition of one or more modules to an existing system requires careful planning.

In centralized power systems in a power cabinet or bay, rectifiers and additional power cabinets or bays may be added as required. If batteries are part of the upgraded system or a centralized power scheme is to be used, rectifier compatibility must be considered. In general, the preferred solution for upgrading power is to install or expand an external DC power plant, such as a DC Rectifer.

For detailed information about system upgrades, see *Avaya Communication Server 1000M and Meridian 1 Large System Upgrade Overview, (NN43021-458)*.

Chapter 9: Selecting proper wire size

Contents

Introduction on page 133

Typical wire values on page 133

Metric conversion on page 134

Calculating wire size on page 135

Sense lead wire size on page 136

Input wire size on page 136

Introduction

This section provides guidelines for determining wire gauges to connect a pedestal to a rectifier, DC distribution panel, or other external power equipment.

Typical wire values

Table 17: Wire characteristics on page 133 lists typical wire sizes in AWG and circular mils for a given maximum current. Table 18: Maximum allowable voltage drops on page 134 lists maximum allowable voltage drops for DC power system conductors.

Table 17: Wire characteristics

Wire gauge (AWG)	Circular mils	Maximum amperes
4	41 750	90
6	26 250	65
8	16 510	50
10	10 380	35

Wire gauge (AWG)	Circular mils	Maximum amperes
12	6530	25

Note:

Maximum amperage is affected by many factors, including temperature and insulation. Consult a wire handbook for precise tables.

Note:

Although gauges smaller than 8 AWG are shown in this table for reference, it is not recommended that sizes smaller than 8 AWG be used for any of the conductors listed in Table 18: Maximum allowable voltage drops on page 134.

Table 18: Maximum allowable voltage drops

Conductor	From	To	Allowable voltage drop (max)
– Battery	Pedestal	– Distribution discharge	1.00
+ Battery return	Pedestal	+ Distribution ground	1.00
– Battery	Distribution	– Battery terminal	0.25
+ Battery return	Distribution	+ Battery terminal	0.25
– Battery	Rectifier	– Distribution charge	0.50
+ Battery return	Rectifier	+ Distribution ground	0.50

Note:

"Distribution" means the DC power distribution panel (box).

Metric conversion

AWG measurements are not directly related to European Industry standard metric measurements. The following table provides guidance when converting from the AWG system to the metric system for the most commonly used power and ground conductor cables.

Table 19: Metric wire conversion

AWG Number	Industry standard Nominal (sq mm)	Resistance at 20 deg.C. (Ohm/100m)
2	35	0.05
4	25	0.08
6	16	0.13

AWG Number	Industry standard Nominal (sq mm)	Resistance at 20 deg.C. (Ohm/ 100m)
8	10	0.20
10	6	0.33
12	4	0.63
14	2.5	1.00
16	1.5	1.40
18	1	2.00
20	0.75	2.90
22	0.5	4.60

Calculating wire size

Using the maximum current in a conductor, determine the length that the conductor must be to meet the required maximum voltage drop. When you know the current, distance, and allowable voltage drop for a specific conductor, calculate the minimum wire size using the following formula:

$$\mathrm{CM} = \frac{11.1 \times \mathrm{I} \times \mathrm{D}}{\mathrm{V}}$$

- CM = wire size required in circular mils
- I = current in amperes (use the maximum expected)
- D = distance in feet (to convert meters to feet, divide by 0.3048)
- V = maximum allowable voltage drop

Caution:

Although the voltage drops listed in Table 18: Maximum allowable voltage drops on page 134 are the maximum drops allowed, the insulation and temperature rating versus current often dictates a wire size that creates smaller voltage drops on short lengths. After using the formula, check the wire tables to make sure the temperature rise is acceptable.

The following examples show wire size calculations using the formula given above.

Example 1

A battery or battery return conductor from a DC distribution panel to a pedestal is 11.0 m (36 ft) long and must carry a maximum of 30 A with voltage drop of no more than 1 V:

$$CM = \frac{11.1 \times 30 \times 36}{1} = 11,988$$

Choosing a standard gauge equal to or larger than this wire size requires #8 AWG, which has a cross-section of 16 510 circular mils.

Example 2

A battery or battery return conductor from a DC distribution panel to the battery is 7.6 m (25 ft) long and must carry a maximum of 35 A:

$$CM = \frac{11.1 \times 35 \times 25}{0.25} = 38850$$

Choosing a standard gauge equal to or larger than this wire size requires #4 AWG, which has a cross-section of 41 740 circular mils.

Sense lead wire size

When sense leads are required, the loop resistance of the wire used to connect the ± sense terminals at the rectifiers or DC distribution panel to the ± terminals of the batteries must not exceed 2.5 ohms.

Input wire size

Table 20: Pedestal wire gauge requirements with two 30 A feeds (five wires) on page 136 provides a means for determining the size of wire used to connect the distribution box and the pedestal. A maximum total voltage drop of two volts is allowed between the pedestal and the external power equipment. Table 20: Pedestal wire gauge requirements with two 30 A feeds (five wires) on page 136 lists cable sizes that give acceptable voltage drops for a given cable length, and those that do not..

Table 20: Pedestal wire gauge requirements with two 30 A feeds (five wires)

Length	#8 AWG	#6 AWG	Single #4 AWG	Double #4 AWG
0–3 m (10 ft)	Yes	Yes	Yes	Yes
3–6 m (20 ft)	Yes	Yes	Yes	Yes
6–9 m (30 ft)	Yes	Yes	Yes	Yes

Length	#8 AWG	#6 AWG	Single #4 AWG	Double #4 AWG
9–12 m (40 ft)	Yes	Yes	Yes	Yes
12–15 m (50 ft)	Yes	Yes	Yes	Yes
15–18 m (60 ft)	No	Yes	Yes	Yes
18–21 m (70 ft)	No	Yes	Yes	Yes
21–24 m (80 ft)	No	Yes	Yes	Yes
24–27 m (90 ft)	No	No	Yes	Yes
27–30 m (100 ft)	No	No	Yes	Yes
30–60 m (200 ft)	No	No	No	Yes
over 60 m (200 ft)	No	No	No	No
Two 30 A feeds are typically adequate for a column with four modules (five wires total — two 30 A feed pairs, BAT(–) and BATRTN(+), plus logic return LRTN(+). Legend: Yes: Wire size is adequate for the distance. No: Wire size has too high a voltage drop and is inadequate for the distance.				

Chapter 10: Preparing a system installation plan

Contents

This chapter contains the following topics:

Introduction on page 139

Creating an installation plan on page 140

Fire, security, and safety requirements on page 142

Equipment room requirements on page 144

Grounding and power requirements on page 151

Cable requirements on page 167

Preparing a floor plan on page 170

Estimating floor loading on page 174

Creating a building cable plan on page 175

Preparing for delivery on page 179

Preparing for installation on page 180

Introduction

Warning:

Before a Large 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 may need additional space for a maintenance and technician area.
- Location:
  - The location should be convenient for equipment delivery and close to related work areas. You must consider the location of related equipment (such as the distribution frame and batteries) and the cable limitations when selecting the site.
- 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 21: Installation plan outline on page 140 as a guide for preparing a detailed installation plan.

Table 21: Installation plan outline

Procedure	Requirements
Researching site requirements	• Determine fire, security, and safety requirements • Determine equipment room requirements

Procedure	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 22: Milestone chart on page 141 lists typical activities included in a milestone chart. For a complex site, you must create a more detailed chart.

Table 22: 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.
11	Complete construction and ensure that grounding and power are in place.
12	Test air conditioning and heating systems.

Task	Action
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 may 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:

1. Sprinkler heads
2. Fire and smoke sensing devices
3. 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 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.

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 (for example, 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.

The acoustic noise generated by a column 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.

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

Although operating needs determine the general location of terminal devices, these devices must not be located beyond the maximum distances defined for their interface cards. 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 23: Operating environment on page 146 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 23: Operating environment

Equipment	Temperature and humidity considerations
Large System	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)	10° to 35°C (50° to 95°F), 20% to 80% Relative Humidity
Telephones	Absolute: • 5° 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. If you operate the system above recommended limits (it must remain within absolute limits), be sure to locate disk drive units in one of the lower two modules in a column.

Follow the specifications listed in Table 24: Storage environment on page 146 to store or transport equipment.

Table 24: Storage environment

Equipment	Temperature/humidity considerations
Large System (without disk drive units)	• –50° to 70°C (–58° 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

Equipment	Temperature/humidity considerations
Telephones	• –40° to 70°C (–40° to 158°F) • RH 5% to 95%, noncondensing
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
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.

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

    For systems with reserve power equipment, consult the manufacturer's specifications for recommended operating temperatures.
3. Heat dissipation from a system is estimated in BTUs 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.

4. Table 25: Heat dissipation modules on page 148 shows the maximum power dissipation in the form of heat for each module. The measurements are the same for AC- and DC-powered modules.

Table 25: Heat dissipation modules

Module	Heat dissipation	
	Watts	Btu/hr
NT4N41 Core/Network	360	1230
NT8D35 Network	240	820
NT8D37 Intelligent Peripheral Equipment	460	1569
NTDW66 CP PM Signaling Server	30	102
NTDU97 Signaling Server (HP DL320 G4)	583	1990
NTDU99 Signaling Server (IBM x306m)	300	1024
NTDW40 Signaling Server (IBM x3350)	400	1365
NTDW41 Signaling Server (Dell R300)	400	1365
700501181 Common Server (HP DL360 G7)	175	592
Thermal load (Btu/hr) = total power dissipation (watts) × 3.41		

5. Table 26: Heat dissipation rectifiers on page 148 shows the maximum heat dissipation for DC-power rectifiers supported by Avaya.

Table 26: Heat dissipation rectifiers

Rectifier	Heat dissipation	
	Watts	Btu/hr
NT5C06 25 A	130	444
NT6D52 30 A	580	1980
A0354954 100 A	580	1980
MFA150 25 A	150	512
Thermal load (Btu/hr) = total power dissipation (watts) × 3.41		

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.

IEEE Standard 142 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. See *Avaya Communication Server 1000M and Meridian 1 Large System Installation and Commissioning, (NN43021-310)* for detailed instructions on installing earthquake bracing.

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

This section describes isolated and nonisolated ground topologies, commercial power source, auxiliary power, and power failure transfer unit (PFTU) requirements. If there is a conflict between information in this chapter and a local or national code, follow the code.

Grounding

Voltage:

DANGER OF ELECTRIC SHOCK

If you fail to follow grounding procedures, the installation can be unsafe for personnel, unprotected from lightning or power transients, subject to service interruptions, and subject to degraded performance.

Power and ground must originate from the supply service (equipment room service panel or transformer), where the ground conductor and the neutral conductor connect and are referenced to the main building ground. All power feeds should contain a separate safety conductor (green wire).

Important:

IMPORTANT!

Do not use the main building ground directly as the ground reference for the system.

The equipment service panel must be located in the equipment room. This service panel must not service lighting, air conditioning, heating, generators, or motors. Avaya strongly recommends that supply conductors be dedicated and uninterrupted from a building primary source to the dedicated equipment room service panel.

Power is supplied to the service panel by a power transformer. The transformer typically provides secondary voltages of 208/120 V three-phase four-wire "wye" service, 240/120 V single-phase four-wire "delta" service, or 240/120 V single-phase three-wire service. Collectively, these secondary voltages are referred to as "nominal 208/240 V AC" throughout system documentation.

A dedicated power transformer for the system and associated auxiliary and telephone operating company interface equipment is preferred; however, a shared transformer or distribution is acceptable. (Figure 47: Dedicated transformer in an isolated ground system on page 156 through Figure 50: Shared distribution in a nonisolated ground system on page 159 starting on Figure 47: Dedicated transformer in an isolated ground system on page 156 illustrate the differences between dedicated and shared distribution.)

Warning:

Do not use ground fault circuit interrupt (GFCI) devices on system AC power feeds

Single point ground (SPG)

The system requires a single point ground (SPG) topology for all equipment and all associated auxiliary equipment.

The system has several types of grounds and several types of signal returns that are generally referred to as "grounds."

- In AC systems, there is a logic return (LR or LRTN) and a green wire frame ground, called the AC equipment ground (ACEG), that is typically part of the input power cord.
- In DC systems, there is a logic return (LR or LRTN) and a battery return (RTN), as well as an AC equipment ground (ACEG) green wire on the input to the rectifier(s).
- All systems must have an external hardwired frame ground connection (also called the personal hazard safety ground). The frame ground is connected internally to the ACEG green wire, but because it is hardwired it ensures that the equipment has a ground connection even if the system is "unplugged."
- External Communications wiring that meet the requirements as stipulated in NEC Article 800-30 FPN 4 require the use of lightning protection. The cable sheaths, and protection grounds must be installed per NEC Article 800 - 33, and Article 800 - 40 (b).

For SPG topology, each of these grounds from each of the columns, must terminate at a single connection point before attaching to the actual ground reference at the service panel or transformer. Physically, the SPG is usually a copper bar or plate (referred to as a "bus"). In its simplest form, the SPG (the single connection point) can be an isolated ground bus or ACEG bus in the service panel or transformer.

In some conditions, a logic return equalizing (LRE) bus is needed. Multiple-column systems, for example, often require an LRE bus as a ground connection point. The LRE serves as the point where the logic return (LR or LRTN) wires from different columns are consolidated before connecting to the SPG.

Two LRE assemblies are available from Avaya:

- NT6D5304 Ground Bus/LRE – Small (provides up to nine connections); typically used with AC-powered systems
- NT6D5303 Ground Bus/LRE – Large (provides up to 48 connections); typically used with DC-powered systems

SPG requirements

Follow these requirements for the SPG:

- All ground conductors must be identified according to local codes and terminated permanently.
- Terminations must be accessible for inspection and maintenance during the life of the installation.
- All grounding conductors must be continuous, with no splices or junctions, tagged "Do not remove or disconnect," and insulated against contact with foreign grounds.
- Grounding conductors must be no load, noncurrent-carrying cables, under normal operating conditions.
- The ground interface, in a steel-framed building, must have a single connecting reference, located at the service panel, to the building steel on the same floor as the system (or within one floor).

Avaya does not recommend the use of building steel as an integral part of the ground system. The building steel is a reference point only.

The DC resistance of the system ground conductor, which runs from the system to the main building ground, must be as close to zero as possible. The maximum total resistance on all runs within the building must not exceed 0.5 ohms.

All voice and data lines that run outside to the building, leaving or entering the system, must have fault protectors that connect directly to an approved ground. Fault protectors provide protection from external faults and transients on data lines. Refer to the 800 section of the NEC Handbook, 1996 edition or later, for what constitutes an approved ground.

To meet system requirements for an SPG:

- Installation must adhere to the SPG requirements.
- The building ground must meet country-specific regulations (example: in the U.S., the National Electrical Code (NEC) regulations must be followed, and in Canada, the Canadian Electrical Code (CEC)).
- Use the proper wire size for the system ground reference conductor.

Isolated and nonisolated ground

You can install the system with an isolated or nonisolated ground topology. Avaya strongly recommends using an isolated ground for grounding system integrity. Use nonisolated ground systems only where they are required by code.

In an isolated ground system, the dedicated isolated ground bus bar in the service panel serves as the ground window. It is used for all AC safety grounds and logic returns. It also accommodates a conductor that references the (+) battery bus in DC systems.

In addition, one or more isolated LREs can be located outside of the service panel, but they must connect to ground exclusively through the isolated ground bus.

Isolated IG-L6-20 or IG-L6-30 orange receptacles are used with an isolated ground system. All ground wiring for isolated ground receptacles must terminate on the dedicated isolated ground bus according to applicable codes.

In a nonisolated ground system, the ACEG connects to the metal panel, and any associated conduit can also contact various structural metal. Because this ground alone is not adequate for the system, a dedicated ground conductor connected to the main building ground is used for the main ground window to terminate logic returns and reference the (+) battery bus. Frame grounds connect to the ACEG.

nonisolated L6-20 or L6-30 brown receptacles are used with a nonisolated ground system.

For more detailed information about receptacles, see Commercial power source on page 159.

All circuit breakers must be clearly labeled in both isolated and nonisolated ground AC panels.

Figure 47: Dedicated transformer in an isolated ground system on page 156 through Figure 50: Shared distribution in a nonisolated ground system on page 159 starting on Figure 47: Dedicated transformer in an isolated ground system on page 156 illustrate the differences between (a) dedicated and shared distribution, and (b) isolated and nonisolated ground systems.

The following notes apply to Figure 47: Dedicated transformer in an isolated ground system on page 156 through Figure 50: Shared distribution in a nonisolated ground system on page 159.

Run the ground conductor in the same conduit with the phase and neutral conductors. Use the appropriate NEC table to determine the correct wire size.

Use of an isolation transformer is recommended. Locate it as close as possible to the AC panel.

You must bond the Ground electrode conductor to a recognized ground, such as a vertical ground riser or a building principal ground. Keep it at a low impedance and do not run it in a conduit. Ground in accordance with the NEC/CEC guidelines.

This conductor may not be smaller than number 6 AWG.

Locate the dedicated equipment service panel in the equipment room.

Amperage level depends on the equipment being fed; see the *Avaya Communication Server 1000M and Meridian 1 Large System Installation and Commissioning, (NN43021-310)*.

For AC systems, this goes to the Logic Return Equalizer (may not be required where enough terminations exist on the IG bus). For DC systems, this goes to the DC ground reference.

Bond Telco/OSP shields, bonds, and protection at an approved reference per NEC Article 800 and CEC Article 10-1000, and Appendix B 36-310 (9). Do not bond them at the LRE or Service Panel.

It is required that all 120 VAC service drops in the equipment room have IG-type receptacles. Each receptacle must have an individual hot, neutral, and IG ground conductor run in the same conduit (NEC 250-74 Exception 4, CEC 10-906(8). Some local codes require an additional bonding lead to bond the outlet box back to the frame panel.

Label circuits at both ends in accordance with NEC 110-2/CEC guidelines. Identify NEMA numbers for IG-type receptacles at the panel and outlet as follows:

- 120V @ 15A = IG.5.15
- 208V @ 20A = IG.L6.20
- 208V @ 30A = IG.L6.30

In Canada, it may be required that the IG ground bus be bonded to the panel frame.

Refer to Auxiliary power on page 162 for more information.

An alternate earthing electrode, if required, must be installed at a minimum of 1.8 m (6 ft) from the building earth reference.

If you use PVC conduit, a dirty grounding conductor may be required.

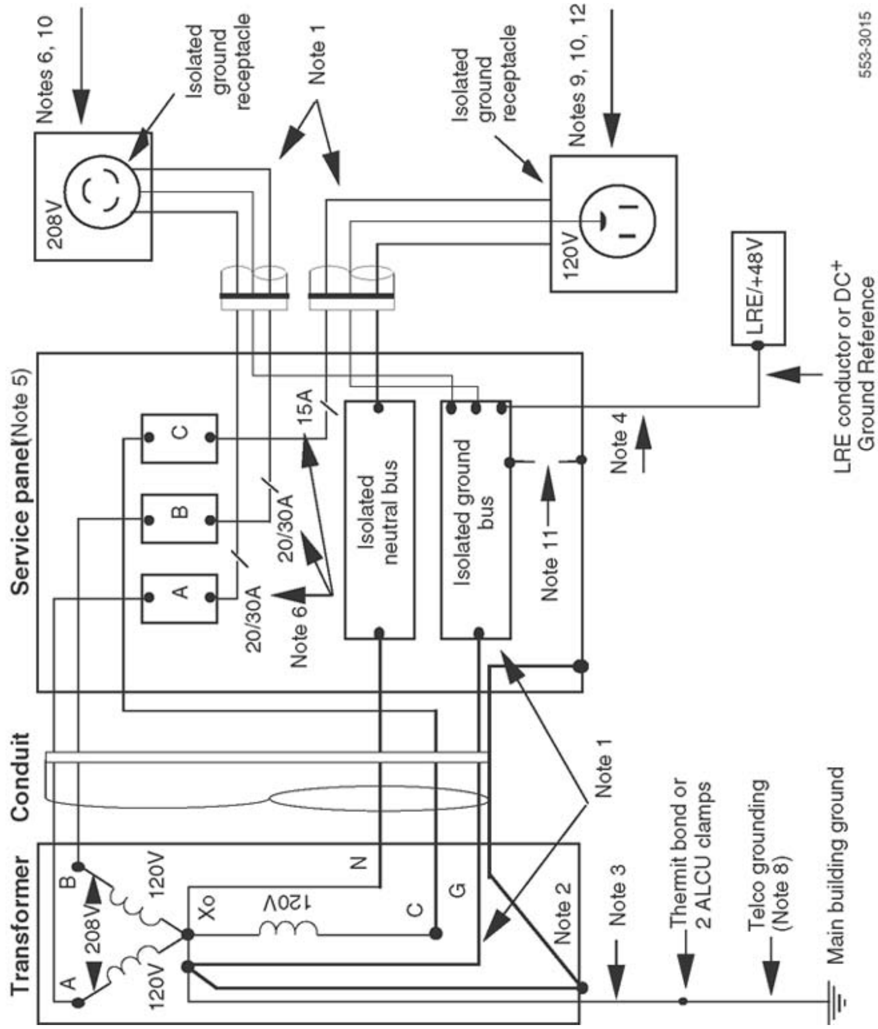

Figure 47: Dedicated transformer in an isolated ground system

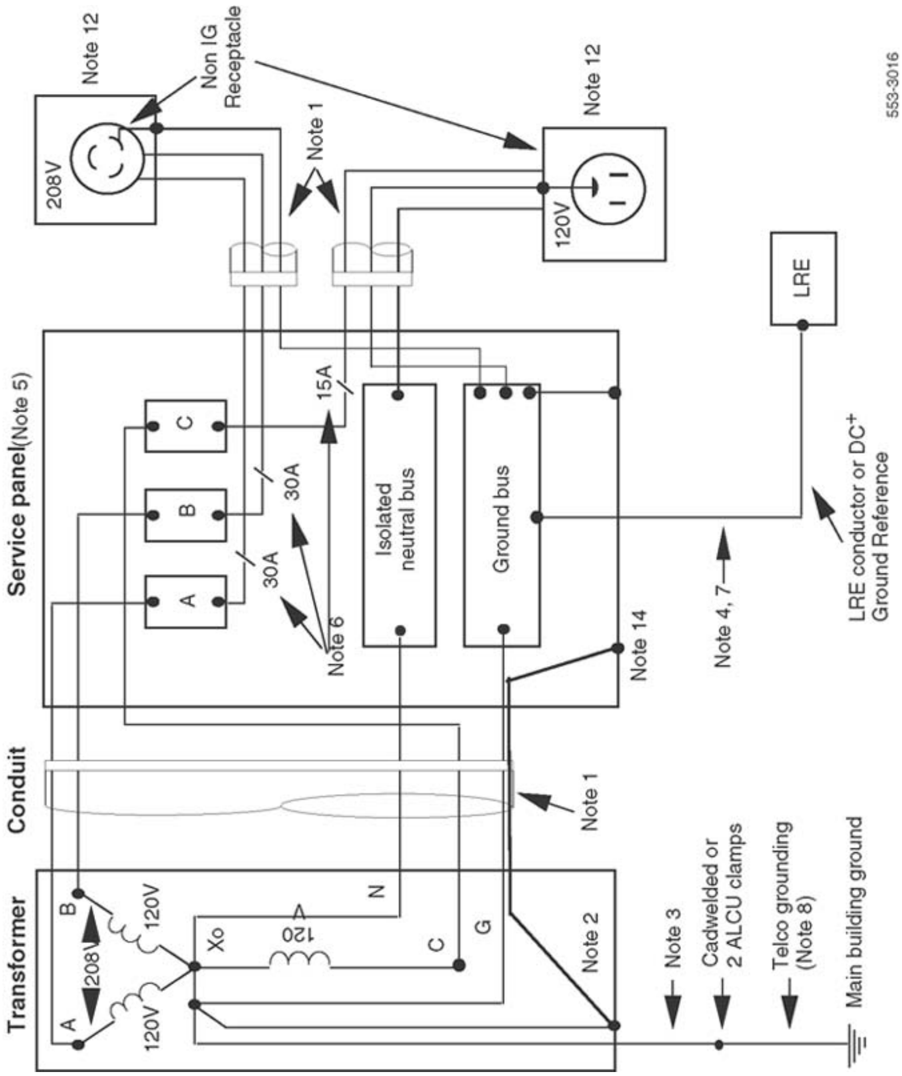

Figure 48: Dedicated transformer in a nonisolated ground system

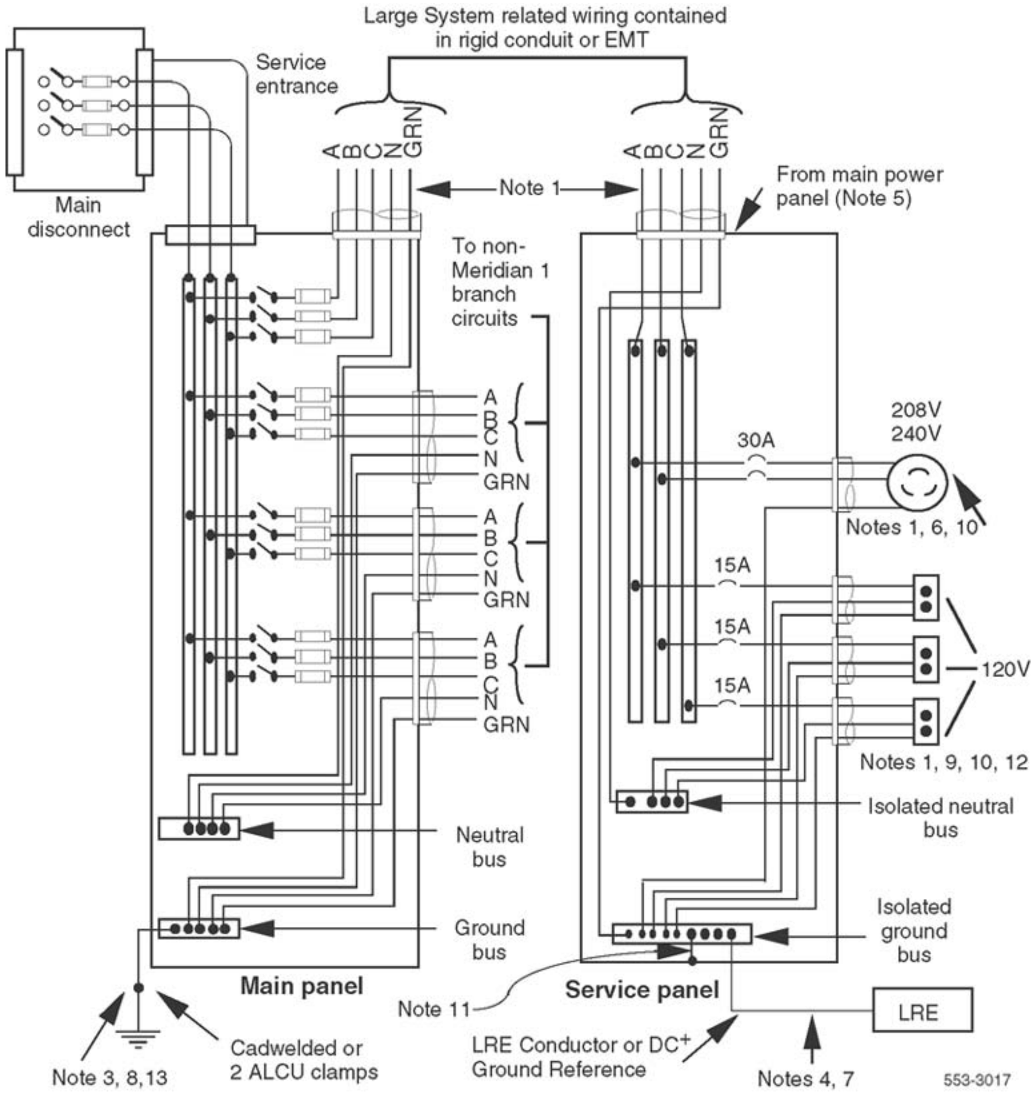

Figure 49: Shared distribution in an isolated ground system

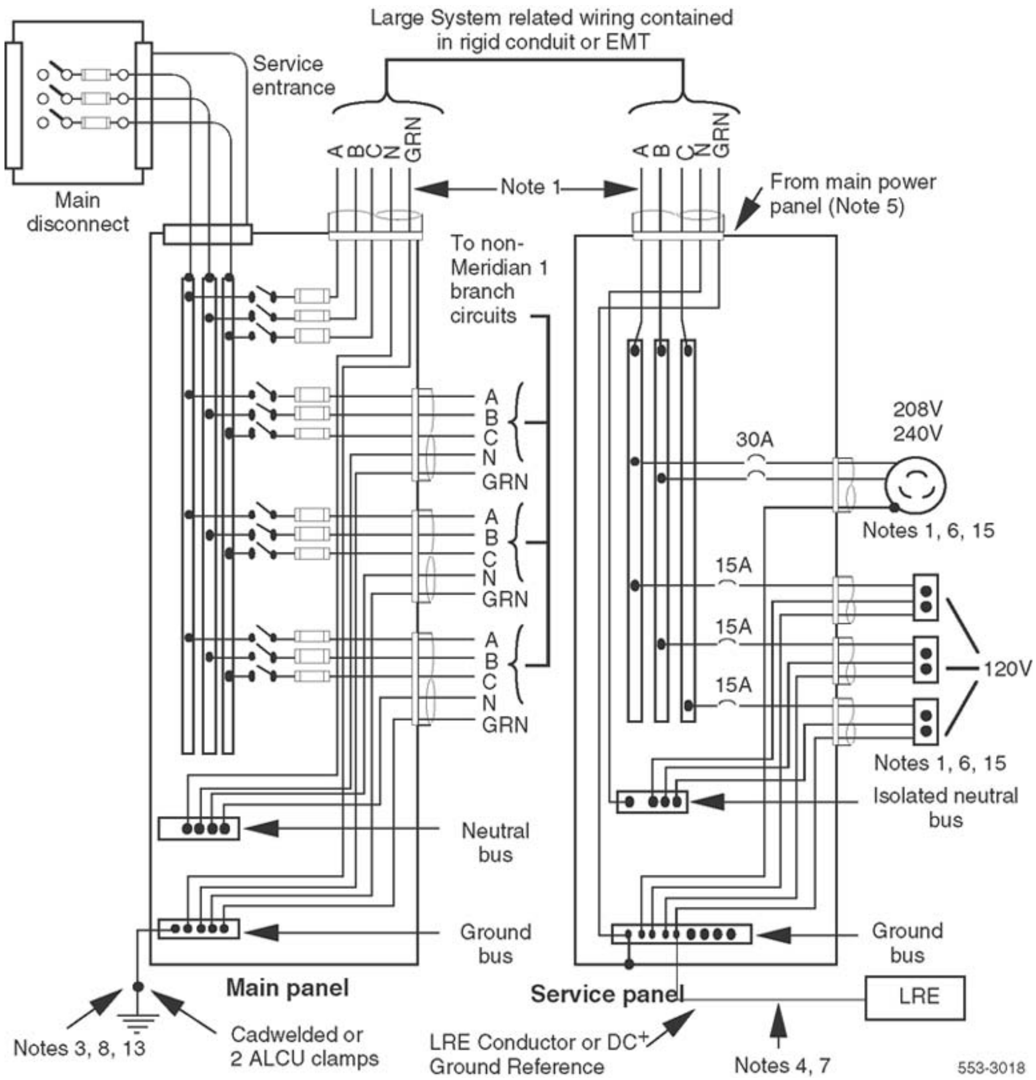

Figure 50: Shared distribution in a nonisolated ground system

Commercial power source

The commercial power source is the main AC utility power feed, which is required for both AC- or DC-powered systems. For AC systems, this power source connects directly to the system. For DC systems, this power source connects to the rectifiers, which convert the AC voltage to –48 V DC voltage for distribution to the system.

In North America, the power supplied can be either 208 V AC or 240 V AC nominal. Three-phase power is not required, but single power feeds from alternate phases (phase-to-phase wiring) are normal practice where three-phase power is available.

Table 27: AC input specifications for AC-powered systems on page 160 lists the input power required from the commercial power source for AC-powered systems. As shown, any voltage in the range of 180 V to 250 V is acceptable.

Table 27: AC input specifications for AC-powered systems

Input	Minimum	Nominal	Maximum
Voltage (V AC) at pedestal	180	208/240	250
Frequency (Hz)	47	50/60	63
Distortion on voltage sine wave: 5% total harmonic distortion (THD), 3% any single harmonic.			

Table 28: Transient tolerance for AC-powered systems on page 160 shows the transient tolerance for abnormally high- and low-line conditions for module power supplies in AC-powered systems. When subjected to these transients, the power supplies continue to maintain their outputs within their specified operating limits. Spikes and notches are defined in terms of 0.5 and 0.25 cycle power disturbances. Surges and sags tend to be temporary changes in the nominal AC voltage, sometimes over several 60 Hz cycles.

Table 28: Transient tolerance for AC-powered systems

Transient	Amplitude	Duration
High-voltage conditions:		
Spikes	815 V AC 408 to 815 V AC	up to 4.16 ms 4.17 to 8.33 ms
Surges	288 V AC 276 V AC	8.34 to 50 ms 51 to 500 ms
Low-voltage conditions:		
Notches	0 V 0 to 206 V	up to 4.16 ms 4.17 to 8.33 ms
Sags	146 V 166 V	8.34 to 50 ms 51 to 500 ms
All transients are applied at the peak of the AC waveform.		

The specifications in Table 28: Transient tolerance for AC-powered systems on page 160 are derived from NEC and various telephone operating company specifications. These specifications are based on power disturbances that have been measured or observed, or that can be expected to commonly occur. Therefore, these specifications for transient tolerance are the minimum requirements that the equipment must meet.

The "hold-up" time specification for an AC module power supply is 20 ms at full load, when measured at the peak of the input voltage waveform and with nominal input of 208 V AC. Hold-up time is the time from the removal of the AC input voltage to the time when any one output voltage drops below its specified operating limit. At less than full load, the hold-up time is greater.

The hold-up specification exceeds the low-voltage transient specifications listed in Table 28: Transient tolerance for AC-powered systems on page 160 above.

Table 29: Service receptacle requirements on page 161 lists the National Electrical Manufacturer's Association (NEMA) numbers for acceptable commercial power service receptacles.

Table 29: Service receptacle requirements

<table>
<tr><th>Receptacles</th><th>Isolated</th><th>nonisolated</th><th>Used with</th></tr>
<tr><td>208/240 V at 20 A</td><td>IG-L6-20</td><td>L6-20</td><td>NT6D52 rectifier</td></tr>
<tr><td>208/240 V at 30 A</td><td>IG-L6-30</td><td>L6-30</td><td>AC systems</td></tr>
<tr><td rowspan="4">208/240 V at 30 A</td><td rowspan="4" colspan="2">hardwired</td><td>NT5C03 rectifier</td></tr>
<tr><td>A0354954 rectifier</td></tr>
<tr><td>NT5C07 rectifier</td></tr>
<tr><td>NT5C06 rectifier (MFA 150)</td></tr>
</table>

Power conditioning

The term "power conditioner" refers to a variety of power protection or power quality improvement devices, such as low-pass filters, surge arrestors, line voltage regulators, and isolation transformers. Some of these devices reduce noise on the commercial power feed, and others help prevent power line spikes and surges. Many uninterruptible power supply (UPS) systems, in addition to providing reserve power for AC-powered systems, provide conditioning and protection during normal operation.

If the quality of the commercial power meets the specifications listed in this document, you do not need power conditioning equipment. If you want protection beyond the transient specifications listed, supplemental power devices can be helpful. However, carefully evaluate the specifications for the power protection equipment to be sure the equipment provides the type of protection that you want.

Power conditioning equipment of any sort is not a substitute for proper system grounding. As emphasized throughout this document, an SPG topology must be maintained for the system and all directly connected switchroom equipment. If you use supplemental protection equipment, you must install it in series with the commercial power feed to the system, without altering the overall grounding scheme.

Auxiliary power

Terminal devices located in the equipment room require local power. Power for these devices must be wired and fused independently from all other receptacles, labeled at the service panel (to prevent unauthorized power interruption), and referenced to the same interface point on the building system ground as the service panel ground.

Auxiliary power in the equipment room can be supplied by isolated or nonisolated service receptacles, but the receptacles must match the grounding for the system. In other words, if the system has an isolated ground topology, the receptacles must also be isolated. You can use the A0367916 Auxiliary –48 V Power Supply as a general purpose power supply for terminal devices (as well as supplying power to PFTUs). All 120 V circuits in the equipment room must have individual hot, neutral, and ground conductors.

If auxiliary equipment using an RS-232 interface is too remote to be powered from the service panel, a modem or fiber link is required for ground isolation. Failure to provide this isolation defeats the SPG required by the system.

Existing powering and grounding on some sites can make it difficult to ensure that the local power grounding is referenced to the same potential as the system ground. In addition, local power grounding can form part of a common grounding network that is subject to noise from external sources. Under these conditions, where locally powered terminals and equipment connect directly to the system through DC-coupled links sharing a common ground, incidental ground loops can form and inject noise into the system.

Where you suspect ground related problems, and you have eliminated other sources of the problem, isolate the auxiliary equipment from the system. The best way to do this depends on the individual installation and local practices, but a few possibilities are listed here:

- Connect the auxiliary equipment to the system through an opto coupler isolation device.
- Connect the auxiliary equipment to the system through fiber-optic links.
- Use teletypewriters (TTYs) configured in the 20 mA loop current mode (such as current loop adapters).
- Use isolation modems configured back-to-back. (Do not reference modems on the system side to the AC ground.)

Isolated service receptacles

For auxiliary power receptacles in isolated ground systems, use 120 V, 60 Hz, 15 A, individually fused, isolated ground receptacles terminating on nonlocking type IG-5-15 receptacles (such as Hubbell, Cat. No. IG-5262, 2-pole, 3-wire, orange duplex receptacles). Use a green conductor for extending the safety ground, and wire it according to the isolated ground specifications. (This requirement is based on safety concerns and exceeds NEC and CEC requirements.)

Outlets must comply with NEC 250-74 Exception 4. Route grounding conductors with the phase conductors (NEC 300-20). All ground wiring must terminate on the dedicated isolated ground bus according to applicable codes (NEC 384-27).

Nonisolated service receptacles

For auxiliary power receptacles in nonisolated ground systems, use 120 V, 60 Hz, 15 A, individually-fused receptacles terminating on nonlocking type 5-15 receptacles.

Power options

Two power options are available:

1. AC-powered systems with or without reserve (backup) power
2. DC-powered systems with or without reserve (backup) power

In any configuration, you can route power connections to the system through the floor or along overhead racks.

AC-powered systems

In an AC-powered system, commercial power voltage is brought directly into the power distribution unit (PDU) in the pedestal. If reserve power is required, install an uninterruptible power source (UPS), along with its associated batteries (which may be internal or external to the unit), in series with the AC power source.

See the manufacturer's specifications for details on the storage and operating environment, especially temperature and humidity ranges, required for proper UPS operation.

AC module power supplies operate at a nominal 208/240 V. The actual input range of AC power supply is 180 to 250 V, so restrapping the power supplies is unnecessary for either 208 V or 240 V power feeds. The 208 V wiring can plug into a 240 V system and vice-versa.

AC-powered systems without reserve power require one input receptacle per column, within 2.4 m (8 ft) of each column's pedestal.

As an alternative to using the power cord and plug, input to the PDU can be wired directly. In this case, #10 AWG conductors routed through 0.75 in. conduit is generally used. The leads connect to the L1, L2, and GND terminations on the field wiring terminal block on the PDU.

Systems that use reserve power plug into the UPS that in turn plugs into the commercial power source. Consult the UPS manufacturer for the receptacle requirements.

DC-powered systems

The external DC power system, generally referred to as the power plant, consists mainly of rectifiers and distribution equipment, and can include batteries for reserve power. DC-powered systems connect to the commercial power source through the rectifiers, which provide –48 V DC to the PDU in the pedestal.

A customer-provided power plant can be used with all DC-powered systems. Refer to the manufacturer's specifications for the power plant requirements.

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."

Power Failure Transfer Unit

A0355200 Power Failure Transfer Units (PFTUs) provide emergency telephone service during commercial power outages or certain system malfunctions. Each PFTU supports up to eight designated telephones that bypass the system and connect the designated telephones directly to the central office (CO) during power failures when activated by the system monitor or when activated manually.

A PFTU always requires a –48 V DC input and a positive return (ground):

- For AC-powered systems:
    - Without reserve power, a separate A0367916 Power Supply –48 V is required. (Up to six PFTUs can be supported by one power supply.) The auxiliary power supply is equipped with a 120 V AC input cord and plug that connects to a properly wired and grounded auxiliary receptacle.
    - With an UPS for reserve power, the auxiliary power supply plugs into an auxiliary 120 V AC output on the UPS.
- For DC-powered systems:
    - A PFTU can be powered from a separately fused auxiliary –48 V feed from the external power system.
    - A separate A0367916 Auxiliary –48 V Power Supply can also be used to power PFTUs in a DC-powered system.

Table 30: Equipment specifications on page 165 provides input power requirements for the PFTU and input and output specifications for the auxiliary power supply.

Table 30: Equipment specifications

Equipment	Input power requirements	Output specifications
PFTU	–40 to –56 V DC 170 mA	—
Auxiliary power supply	90 to 130 V AC at 57 to 63 Hz	–48 V DC, ± 15% at 1.25 A

The PFTU is a wall-mount unit. The auxiliary power supply can be mounted on the floor or wall. PFTU and auxiliary power supply dimensions are given in Table 31: PFTU and auxiliary power supply dimensions on page 165.

Table 31: PFTU and auxiliary power supply dimensions

Equipment	Width		Length		Height		Weight	
	cm	in.	cm	in.	cm	in.	cm	in.
PFTU	12.1	4.75	34.3	13.5	4.1	1.6	1.5	3.3
Auxiliary power supply	12.7	5.00	16.7	6.6	6.4	2.5	1.0	2.2

QUA6 Power Fail Transfer Unit

The QUA6 Power Fail Transfer Unit provides emergency telephone service during commercial power outages or certain system malfunctions. Each QUA6 PFTU supports up to five designated telephones. The PFTU bypasses the system and connects the designated telephones directly to the central office during power failures, when activated by the system monitor, or when activated manually.

Input requirements

The PFTU requires a –48 V DC input and a positive return (ground). The PFTU is powered from a separately fused auxiliary –48V feed from the external power system.

Input requirement: QUA6 PFTU:–42 to –56 V DC at 150 mA nominal

Dimensions and weight

The QUA6 PFTU is a wall-mounted unit and weighs 2 lbs (0.8 kg). The dimensions of the unit are shown in Figure 51: QUA6 PFTU dimensions on page 167.

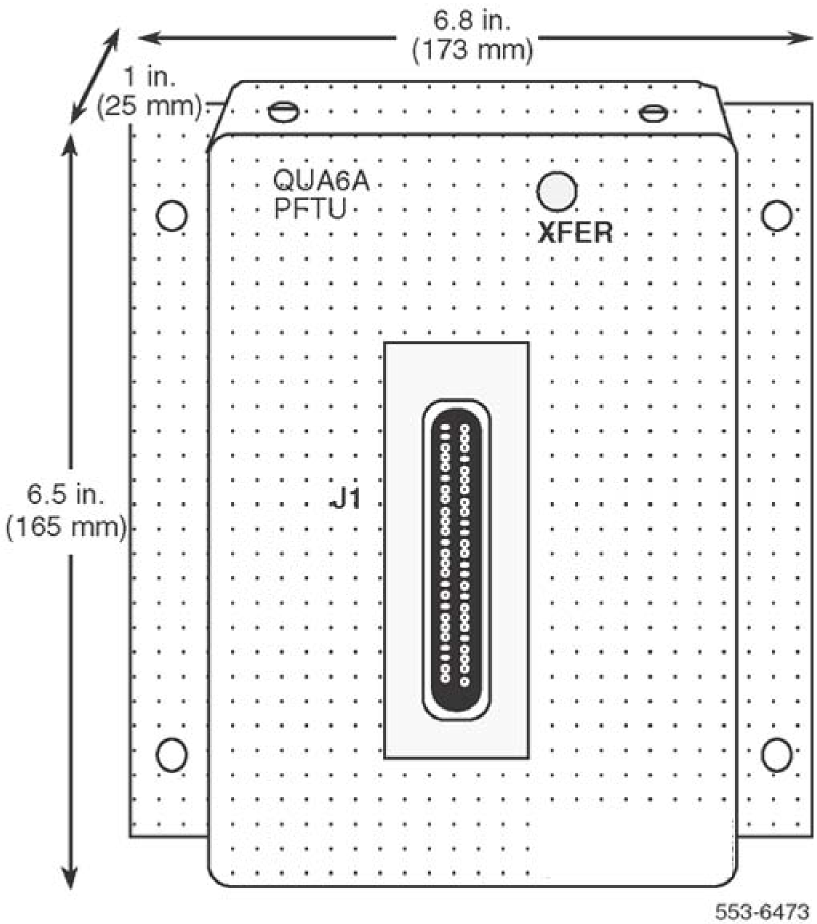

Figure 51: QUA6 PFTU dimensions

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 modules 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.
- Twisted-pair shielded and nonshielded cables: These cables interconnect the trip power monitoring connections between power interface units and the MDF. Typically, a #22 AWG, stranded (Belden type 8408-2 conductor or equivalent) shielded cable is used for trip connections and to connect the system to Power Cabinets. All other connections are serviced by nonshielded, #22 AWG stranded cable.
- Twisted-pair telephone cables: These cables carry analog voice and digitized voice and data information between distribution frames and terminal devices throughout the building. They connect to 8-pin modular jacks located within 2.4 m (8 ft) of each device.

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

System cabling

This section contains information about:

1. Power and ground cables
2. Module cable routing
3. Network to Peripheral Equipment cabling

Power and ground cables

For AC-powered systems, a 2.7 m (9 ft), 3-conductor line cord is supplied, except in areas where conduit is required.

For DC-powered systems equipped with an NT7D10 PDU, wiring is generally run through conduit. For systems equipped with an NT7D67CA PDU, conduit is not required. However, conduit may be used, if preferred or required by local code or practices, and attached to the pedestal at any of three locations. (Rear access is provided by the NT7D0902 Rear Mount Conduit Kit.)

Metallic conduit is used primarily to contain electromagnetic emissions. Where conduit is used, it must provide an end-to-end enclosure for the power wiring.

Metal ducts and raceways usually do not provide electromagnetic containment; they can be used with, but not in place of, conduit.

Module cable routing

Because the cable troughs and spaces on the sides of each module are within the EMI shielding of the system, unshielded cables can be routed in those areas. The corner vertical channels in the rear of the module are outside of the EMI shield. Cables routed in the vertical channels must be shielded, and must enter and exit the EMI-shielded area through I/O panels and adapters.

As space permits, you can route cables in the following ways:

- Horizontally in the cable troughs at the front, rear, and sides of the module

  In a DC-powered module, because there is no module power distribution unit (MPDU), there is room to route cables horizontally from front to rear on the left side (front view) of the module.
- Vertically on the sides of the module
- Vertically in the corner channels in the rear of the module (shielded cables only)

⚠ **Caution:**

Loss of Data

You must route cables as perpendicular as possible to any nearby power cables. Avoid routing cables near power cables if alternate routing is available. (At the rear of the module, cables routed between the I/O panel and the rear cover can be parallel to the power cables because the panel provides EMI shielding.)

Network to Peripheral Equipment cabling

Cabling between the network and Peripheral Equipment runs from the faceplate of an NT8D04 Superloop Network Card to the backplane connectors for an NT8D01 Controller Card in an intelligent Peripheral Equipment (IPE) Module.

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

Preparing a floor plan

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

- The system columns and modules, including planned expansion areas
- The main distribution frame (MDF)
- The service panel
- System terminal, printer, or other terminal devices (such as modems)
- External power equipment (such as rectifiers)
- Any cable racks
- PTFUs and auxiliary power supplies (if either are equipped)
- Space for additional equipment, such as reserve power equipment or auxiliary processors

Follow these guidelines when you plan the equipment room layout:

- The minimum acceptable distance between equipment aisles is 76 cm (30 in.)
- The minimum acceptable distance between the end of the column and walls, and between rows, is 91.r cm (3 ft)
- The minimum acceptable ceiling height is 243.8 cm (8 ft) or greater

Important:

According to the National Fire Code, equipment must be at least 30.5 cm (12 in.) from a sprinkler head. If a system is four modules high with a cable rack, do not place the equipment directly under any sprinkler heads.

When building or upgrading multiple-group systems, you must consider network expansion in the floor plan because network group modules must be collocated. There are several ways to expand the system. One way is to provide space for additional network groups to the left of the CPU modules, and additional Peripheral Equipment (IPE or PE) to the right. Another way is to add Peripheral Equipment modules in a separate row of columns.

Equipment dimensions appear in Table 32: Equipment dimensions on page 171. Figure 52: Communication Server 1000M SG equipment room floor plan on page 172 and Figure 53: Meridian 1 Option 81C CP PIV equipment room floor plan on page 173 starting on Figure 52: Communication Server 1000M SG equipment room floor plan on page 172 illustrate sample equipment room floor plans. These samples may vary from your floor plan, depending on your system needs and the size and arrangement of your equipment room.

Table 32: Equipment dimensions

Equipment	Width		Depth		Height	
	cm	in.	cm	in.	cm	in.
Pedestal	81.3	32.0	66.0	26.0	25.4	10.0
Top cap	81.3	32.0	55.9	22.0	10.2	4.0
Module	81.3	32.0	55.9	22.0	43.2	17.0
One-module column	81.3	32.0	66.0	26.0	78.7	31.0
Two-module column	81.3	32.0	66.0	26.0	121.9	48.0
Three-module column	81.3	32.0	66.0	26.0	165.1	65.0
Four-module column	81.3	32.0	66.0	26.0	208.3	82.0
Note: Multiple-column systems require a 7.6 cm (3 in.) spacer between each column for cable routing and to provide EMI shielding.						

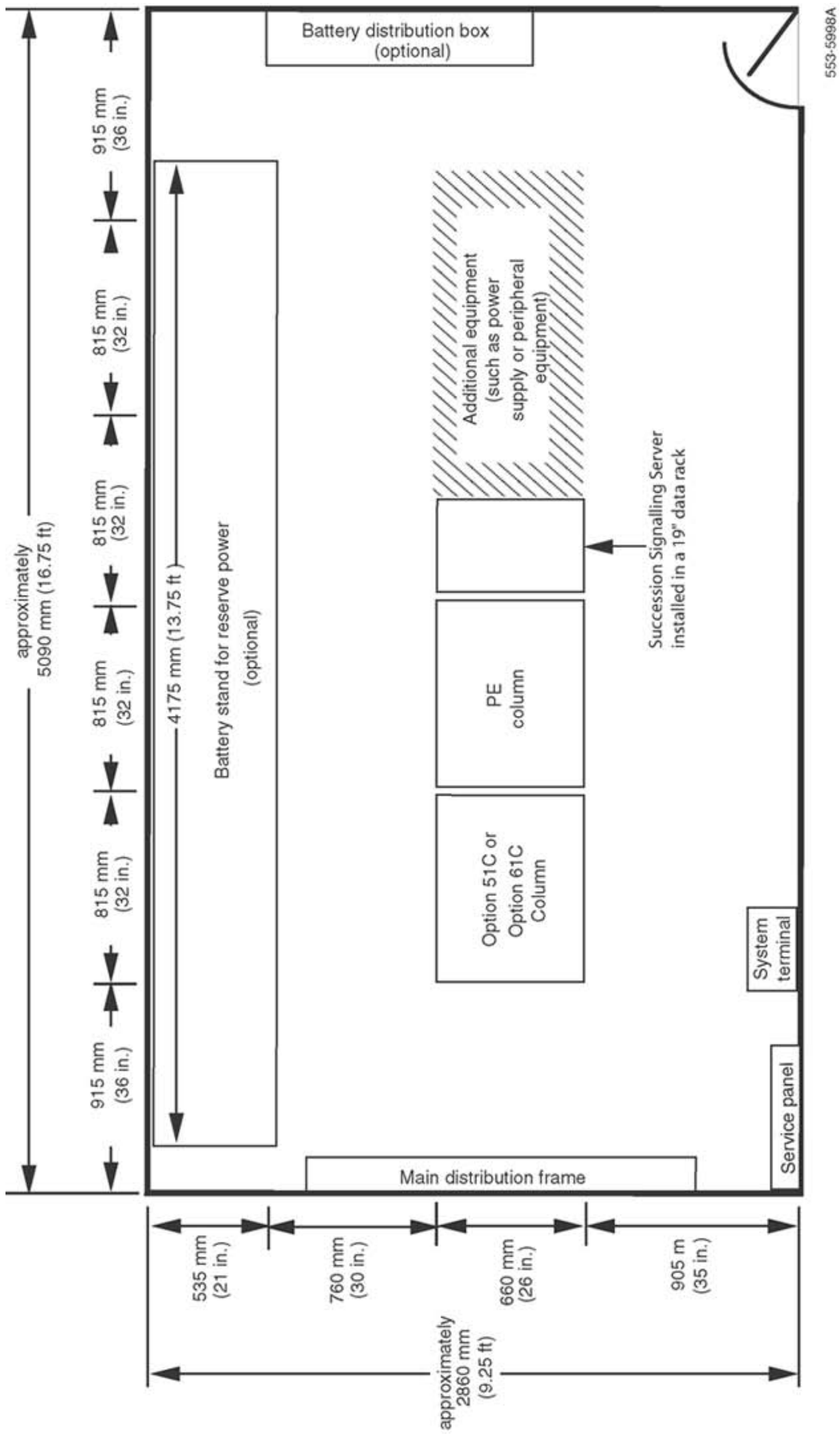

Figure 52: Communication Server 1000M SG equipment room floor plan

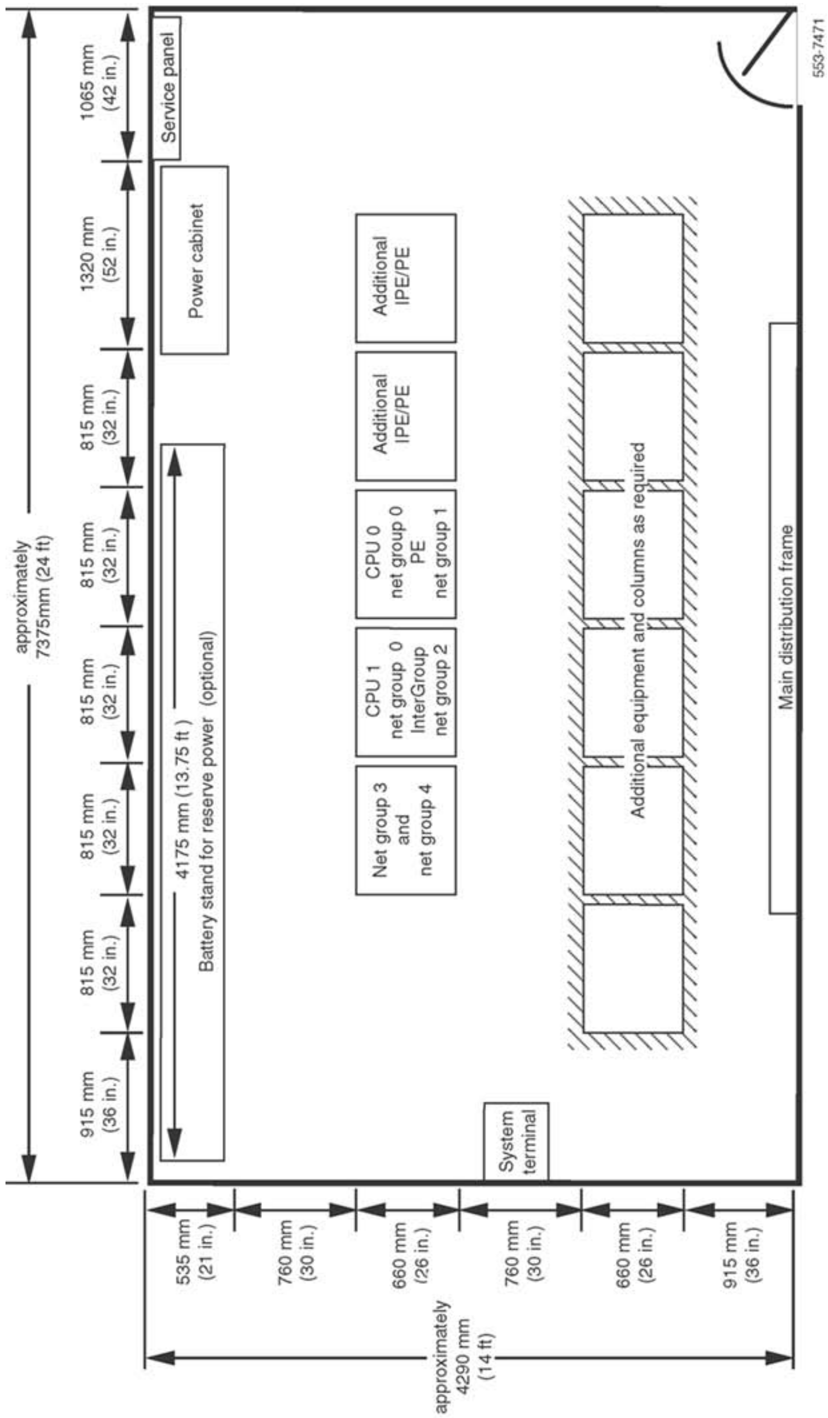

Figure 53: Meridian 1 Option 81C CP PIV equipment room floor plan

Estimating floor loading

You must estimate floor loading to plan module distribution. "Floor loading" is the weight of the system divided by the occupied floor area. "Point loading" is the local pressure exerted by the feet of the system on the floor.

Table 33: Equipment weights on page 174 gives system weights. Table 34: Floor loading estimates on page 174 lists floor load estimates, and assumes fully loaded columns complete with pedestal, maximum circuit card configurations, power supplies, and cables.

Table 33: Equipment weights

Equipment	Weight empty		Weight full	
	kg	lbs	kg	lbs
Pedestal	18.1	40	31.7	70
Top cap	6.8	15	6.8	15
Module	22.7	50	58.9	130
One-module column	N/A	N/A	97.5	215
Two-module column	N/A	N/A	156.5	345
Three-module column	N/A	N/A	215.4	475
Four-module column	N/A	N/A	274.4	605

Table 34: Floor loading estimates

	Floor load		Point load	
Number of modules	lbs/ft2	kPa	lbs/ft2	kPa
One	38.1	1.8	11.0	75.8
Two	60.3	2.8	17.3	119.0
Three	82.4	3.9	23.7	163.4
Four	104.6	5.0	30.0	206.8
The numbers under "Floor load (lbs/ft2) and kPa" are based on a floor area of 0.52 sq m (5.64 sq t) for the system. These numbers do not include the weight of the optional overhead cable rack. The numbers under "Point Load (lbs/in2) and (kPa)" are based on distributing the system weight among four feet, each with an area of 317 sq mm (4.91 sq in.); these numbers do not reflect the use of optional casters.				

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 can not be used for any other wiring.
6. Identify the location of all main and intermediate distribution points.
7. 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.
8. Provide a 16-pair (or 25-pair) cable equipped with an Amphenol-type connector for each attendant console.
9. 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 54: Building cable zones on page 177 illustrates zoning.

   Be sure to leave room for expansion.

Wire routing

To plan wire routing, establish the start and end point of each cable relative to the location of the terminal devices in the building, 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 hat 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 b e 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 if required if conduit is installed.
- EMI:

  Data degradation may occur if wires travel near strong EMI sources. See Electromagnetic and radio frequency interference on page 150 for a description of common interference sources.

Termination points

Once you have determined the wire routing, establish termination points. Cables can terminate at:

1. the MDF (typically in the equipment room)
2. intermediate distribution frames, typically on each floor in telephone utility closets
3. 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 (U.S.A.) 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 55: Sample cable record on page 178 for an example.

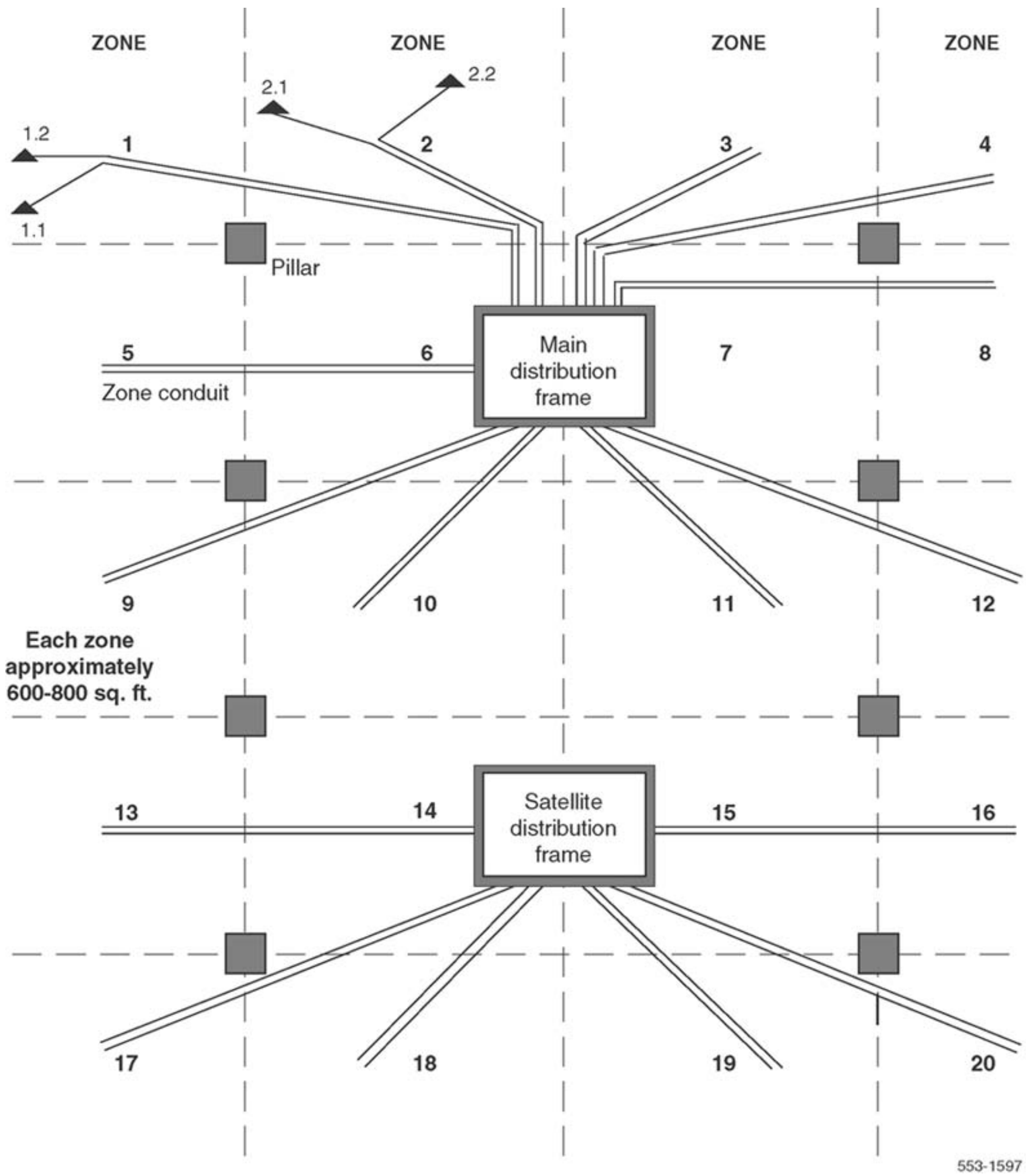

Figure 54: Building cable zones

CABLE RECORD

Customer ____________________________________

Location ______________________________________

Cable ______________________ Binder ____________ Page ____ of ____

DN	TN				NAME	FEATURES / REMARKS	TERMINAL DEVICE	BLOCKS	COLOR
	M	S	C	U				DF / HOUSE	
									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 55: Sample cable record

Preparing for delivery

When preparing for equipment delivery, answer these questions.

1. Has a request been made for equipment delivery?
2. Are transportation arrangements to the premises completed?
3. Is a list of all ordered equipment available on site?
4. Is help needed and available for preparing the equipment room?
5. Are unloading and unpacking facilities and tools available?
6. 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 preinstallation inspections

Obtain any appropriate sign-offs before the site is ready for equipment delivery and installation. Sign-offs 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 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.

Assessing the delivery route

Before the system is delivered, examine and measure the route from the receiving area to the installation area. (See Table 32: Equipment dimensions on page 171 for dimensions.)

These factors must be considered:

1. Size and security of unloading and storage areas
2. Availability and capacity of elevators

3. Number and size of aisles and doors on the route
4. Restrictions, such as bends or obstructions, in halls or at doors
5. Floor loading capacity of unloading, storage, and equipment room areas
6. Number of steps and stairways

   A four-module column is shipped in two segments. One shipping pallet carries the pedestal and three modules. Another shipping pallet carries the fourth module and top cap.

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:

- System columns and modules, including planned expansion areas
- Main distribution frame
- Service panel
- System terminal, printer, or other terminal devices
- External power equipment (such as rectifiers)
- Cable racks
- PFTUs and auxiliary power supplies (if either are equipped)
- Additional equipment such as reserve power equipment or auxiliary processors

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
- 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 and a full set of standard publications, are delivered with each system.

Chapter 11: Design parameters

Contents

This chapter contains the following topics:

Introduction on page 183

System parameters on page 183

Customer parameters on page 184

Console and telephone parameters on page 185

Trunk and route parameters on page 186

ACD feature parameters on page 187

Special feature parameters on page 188

Hardware and capacity parameters on page 189

Memory-related parameters on page 190

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.

System parameters

Table 35: System parameters on page 184 lists system parameters and provides their maximum values.

Table 35: System parameters

System parameters	Maximum value	Comments
Customers	100	
Display messages for background terminal	255	
Input/output ports (e.g., TTYs, printers)	16	Each MSDL counts as one device; a history file counts as one device.
AML/CSL links	16	With MSDL.
TNs	65 536	Software design limit. Actual number of TNs will be constrained by physical capacity, real time, memory, and License limits.
Media Card	256	Maximum number of Media Cards, which includes banks of 32 DSPs on DSP daughter cards.
DSPs	8192	Maximum number of DSPs on a system (256 * 32)

Customer parameters

Table 36: Customer parameters on page 184 lists customer parameters and their maximum values.

Table 36: Customer parameters

Customer parameters	Maximum value	Comments
Tenants	512	
Dial Intercom Groups	2046	
Members per Dial Intercom Group	100	
Ringing Number Pickup groups	4095	Call Pickup Group 0 = no pickup group
Listed Directory Numbers (direct inward dialing only)	6	
DISA DNs	240	

Console and telephone parameters

Table 37: Console and telephone related parameters on page 185 lists console and telephone-related parameters and their maximum values.

Table 37: 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 and 3905 Digital Deskphones Addons • IP Phone 2002 KEM • IP Phone 2004 KEM	1 two-key Key Based Addon (KBA) 1 Display Based Addon (DBA) 1 one-page KEM 2 one-page KEM or 1 two-page KEM	
Protect bcs block length	512	

Trunk and route parameters

Table 38: Trunk and network-related parameters on page 186 lists trunk and network-related parameters and their maximum values.

Table 38: 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 package (400), 16 000 with the package.
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.
ISDN D-channels	64	With MSDL.
ISDN B-channels per D-channel	382	16 T1s with a D-channel and backup D-channel, subject to members per

Trunk/network parameters	Maximum value	Comments
		trunk route limitations and physical limitations.
	359	15 T1s with a single D-channel, subject to members per trunk route limitations and physical limitations.

ACD feature parameters

Table 39: ACD feature parameters on page 187 lists ACD feature parameters and their maximum values.

Table 39: ACD feature parameters

ACD parameters	Maximum value	Comments
ACD DNs and CDNs per customer	- 1000	The ACD-E package required, otherwise the limit is 240.
	- 2000	PKG 411 must be equipped.
Agent positions per DN	- 1200 (Large systems)	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 may limit this further.
AST DNs per telephone	2	
	4	PKG 411 must be equipped.
Number of ACD-ADS customers	5	
Terminals and printers on CCR	8	
Links per VASID	1	

Special feature parameters

Table 40: nonACD feature parameters on page 188 lists nonACD feature parameters and their maximum values.

Table 40: nonACD 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.
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 set 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 max of 20 LTIDs. The max number of LTIDs is limited by

Feature parameters	Maximum value	Comments
		the number of DSLs, memory, and real time.
Mobile Extensions per customer	12 000	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 41: Physical capacity/hardware-related parameters on page 189 lists hardware and capacity parameters and their maximum values.

Table 41: Physical capacity/hardware-related parameters

Physical capacity/hardware parameters	Maximum value (loops)	Comments
XCT cards	64	Provides TDS, CONF, and MFS functionality; requires 2 loops (TDS and MFS share timeslots on one loop, CONF uses the other loop).
Total service and terminal loops: – Avaya Communication Server 1000M (Avaya CS 1000M) SG/PBX 61C – Avaya CS 1000M MG/PBX 81C (FNF, 8 groups)	32 256	Each XNET card requires 4 loops. Each MISP card requires 2 loops.
Digitone receivers	1024	Software design limit.
Multifrequency receivers	255	Software design limit.

Media Cards

Media Cards provide DSP resources. Media Cards do not support the IP Line application.

Media Card is a term used to encompass the Media Card 32-port secure line card, Media Card 32-port line card, and the Media Card 8-port line card.

Media Cards can be assigned to any slot. The slot should be in a nonblocking segment.

Table 42: Media Card capacity

Parameter	Capacity
Meridian 1 Option 61C or Meridian 1 Option 81C	• 10 cards in each IPE cabinet • no more than 3 cards per superloop

Memory-related parameters

Table 43: Memory-related parameters on page 190 lists memory-related parameters and their maximum values.

Table 43: Memory-related parameters

Parameter	Values
Low-priority input buffers • (CP PIV recommended default)	96 – 5000 (3500)
High-priority input buffers • (CP PIV recommended default)	16 – 5000 (3500)
Input buffer size (words)	4
Analog (500/2500-type) telephone, trunk and digital telephone output buffers • (recommended default)	16 – 5000 (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 defaults: • 61C / 81C CP PIV with five or fewer groups expected maximum • 81C CP PIV with six to eight groups expected maximum	80 – 65 000 30 000 35 000
Call registers assigned to AUX	0

Parameter	Values
Number of AML msg call registers	20 – the minimum of 25% of total call registers or 255 (default 25)
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 25% of total call registers (NCR).	

Buffer limits

The buffer limit is the maximum number of CRs 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 43: Memory-related parameters on page 190 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 43: Memory-related parameters on page 190. 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 Large System Call Center using many applications (such as Avaya CallPilot), it would be advisable to set the CSQI/CSQO to a high value (For example, with a Large System Call Center using many applications (up to 25% of total call registers (NCR)).

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 12: System capacities

Contents

This chapter contains the following topics:

Introduction on page 193

Memory size on page 194

Mass storage on page 197

Physical capacity on page 198

Network traffic on page 208

Real-time capacity on page 218

Signaling Server on page 223

Software configuration capacities on page 232

Communication Server 1000M capacities on page 233

Zone/IP Telephony node engineering on page 234

Introduction

This chapter describes the system's primary capacity categories. For each category, this chapter:

- identifies the units in which 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 247 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 311.

The worksheets in Worksheets on page 381 implement the algorithms.

Memory size

Large systems with a Pentium IV processor (CP PIV) use DRAM for both program store and data store.

The following memory configurations are available:

NT4N39AA CP PIV card:

- CP PIV can support 2 DIMM slots for up to 1GB of memory; however,
- CP PIV Pack ships with 512MB of DRAM

Table 44: Avaya CS 1000 memory requirements on page 194 shows the minimum amount of memory required for Avaya Communication Server 1000 (Avaya CS 1000) Release 7.5 software.

Table 44: Avaya CS 1000 memory requirements

System	Flash memory required	DRAM memory required	Total memory
With CP PIV processors			
Communication Server 1000M SG/PBX 61C CP PIV	N/A	512 MB	512 MB
Communication Server 1000M MG/PBX 81C CP PIV	N/A	512 MB	512 MB

Table 45: Maximum memory sizes (MB) on page 194 shows the maximum amount of memory that the processors can support.

Table 45: Maximum memory sizes (MB)

Processor	EPROM	DRAM
CP PIV	N/A	1 GB

Table 46: Recommended maximum call register counts on page 195 gives the maximum call register count recommended for Communication Server 1000 software, so that the system's memory requirements do not exceed the processor's memory capacity.

Sites experiencing memory shortages during an upgrade should check that the call register counts are within the bounds set by this table. Also check that the machine meets the minimum memory requirements listed in Table 44: Avaya CS 1000 memory requirements on page 194.

Table 46: Recommended maximum call register counts

System	Recommended call register count	Memory required (SL-1 words)	Memory required (MByte)
Communication Server 1000M SG Meridian 1 PBX 61C CP PIV	30 000	7 680 000	29.3
Communication Server 1000M MG Meridian 1 PBX 81C CP PIV > 5 groups	35 000	8 960 000	34.18
Call registers are 228 SL-1 words long. One SL-1 word is 4 bytes.			

Table 47: Signaling Server memory on page 195 shows the typical memory configured on each platform. Communication Server 1000 Signaling Servers require a minimum of 2 GB memory. You can upgrade your current CP PM hardware to 2 GB of memory with a Linux Upgrade Kit. The ISP 1100 Signaling Server is not supported.

Table 47: Signaling Server memory

Signaling Server platform	Memory (RAM)
CP PM	2 GB
CP DC	2 GB
HP DL320-G4 or IBM x306m	2 GB
Dell R300 or IBM x3350	4 GB
HP DL360-G7	4 GB

Memory engineering

Current call processors for the Communication Server 1000M 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 at the available memory, create a single set and see how much memory was consumed. Then determine if there is sufficient memory left to create all of the desired sets.

Call register usage

Call register requirements on a system vary with usage and call patterns. In general you want at least 20% more call registers than sets, but this can vary with trunk usage or other features (ACD).

Assumptions:

- Call Register Traffic Factor (CRF) = 1.865
- The formula for calculating the recommended number of call registers depends on traffic load for the system.
- 28 CCS per ACD trunk
- Snacd = (Number of calls overflowed to all target ACD DNs × 2.25) – (Number of calls overflowed to local target ACD DNs × 1.8) (= 0 if the system is not a source node)
- Tnacd = 0.2 × Number of expected calls overflowed from source (= 0 if the system is not a target node)
- ISDN CCS = PRI CCS + BRI CCS
- ISDN penetration factor: p = ISDN CCS ÷ Total Voice Traffic
- ISDN factor: (1 – p)2 + [4 × (1 – p)] × p + (3 × p2)

If Total Voice Traffic > 3000 CCS, then:

Recommended number of call registers = (CRF * 0.071 × Total Voice Traffic) + (0.33 × Number of ACD incoming trunks) + [(Snacd + Tnacd) × 0.03 × ISDN factor]

If Total Voice Traffic < 3000 CCS, then:

Recommended number of call registers = [(Number of system equipped ports – Number of ACD incoming trunks – Number of ACD agent telephones) × 0.94] + {(Number of ACD incoming trunks * 1.21) + [(Snacd + Tnacd) × 0.03]}

A general call register equation would be:

Recommended number of call registers = total ports + (total ports × trunking factor)

trunking factor = (1 – p)^2 + [4 × (1 – p)] × p + (3 × p^2)

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

Mass storage

CP PIV system software installation and database backups are stored on Compact Flash (CF).

CP PIV Compact Flash storage

CP PIV software is stored on an internal 1GB CF that acts as an ATA drive, which is then partitioned into three 305MB partitions. The software is loaded from this drive at start up into DRAM memory to operate.

The software is initially installed onto the system through a faceplate accessible CF (RMD). This is typically a 512MB Card. An installation program will copy the software, keycodes, and customer configuration database onto the on-board CF.

The faceplate CF slot can also be used for customer database backups and log file storage. You can insert a CF card in the faceplate slot to store backups and port blocking memory log files.

CP PIV uses a face-plate CF for backup. This enables CP PIV to archive several databases. Table 48: Projected CF space requirements for SL-1 customer data on page 197 shows the amount of space required on the external media. These are conservative but realistic estimates; not all sites with the given machine type will have databases as large as shown.

Table 48: Projected CF space requirements for SL-1 customer data

System	Estimated size in MB
Meridian 1 PBX 61C, Meridian 1 PBX 61C CP PIV, CS 1000M SG	1.04
Meridian 1 PBX 81C, Meridian 1 PBX 81C CP PIV, CS 1000M MG <= 5 group system	2.92
Meridian 1 PBX 81C, Meridian 1 PBX 81C CP PIV, CS 1000M MG 6-8 group system	4.69

Physical capacity

Resource constraints consist primarily of loop and card slot limitations at the network shelf. From practical experience, running out of PE shelves is rare, particularly for Call Center applications.

This section provides information to calculate physical requirements and capacity considerations for:

- loops (see Loop constraints on page 198)
- card slots (see Card slots on page 202)
- signaling and data links (see Signaling and data links on page 206)

Loop constraints

The maximum number of loops in a network group is 32, including service loops. For practical applications, the number of traffic loops is usually limited to 28, reserving two loops each for TDS and Conference.

Estimate loop requirements separately for the following categories:

- Non-Automatic Call Distribution (non-ACD) telephones and analog trunks on page 199 (N_0)
- Agent sets and ACD analog trunks on page 199 (N_1)
- DTI/PRI trunks on page 200 (N_2)
- Loops for Avaya CallPilot, Music (MUS), and Music Broadcast applications on page 201 (N_3)
- Media Card on page 201 (N_4)

Note on notation

In the calculations that follow, $[\ ]^{+}$ means use the next higher integer, or "round up." For example, $[4]^{+} = 4$ and $[3.1]^{+} = 4$. This document simplifies notations by omitting the "+" at the upper right corner of the bracket. Therefore, [×] means round up × to the next higher integer.

Non-Automatic Call Distribution (non-ACD) telephones and analog trunks

Treat non-ACD telephones and trunks differently from ACD applications for estimating loop requirements. Non-ACD telephone and trunk circuits are equipped in the PE shelf and do not use slots in the Network shelf. Therefore, they are not included in the Network Module Card Slots Calculation.

If there is any doubt about potentially running out of PE slots for a given application (for example, Hotel/Motel environment), review the PE slots to check possible card slot limitations. Since this is a rare occurrence, a calculation procedure is not provided.

For Call Center applications, due to high centi-call seconds (CCS) on circuits (agents or trunks), there is no need to be concerned about physical slot constraints on the PE shelf since real time will be the limiting resource.

When Primary Rate Interface (PRI) trunks are involved in non-ACD applications, they should be treated just like ACD PRI trunks and included in the calculations for both loop and card slot requirements.

The following procedure applies for general and Call Center applications. For IPE XNET loops:

Number of loops for non-ACD set and trunk traffic = N_{0x}= [(No. of sets × 6 + No. of non-ACD trunks × 26) ÷ 875]

The above calculations account for blocking XNET loops.

The default values of 6 CCS per set and 26 CCS per trunk can be replaced by actual numbers for a particular site if they are given. Note that the default trunk traffic assumed for non-ACD application is lower than that of an ACD trunk (28 CCS). The 875 CCS per loop in IPE is derived from superloop capacity of 3500 CCS divided by 4 to obtain the average CCS per loop.

Agent sets and ACD analog trunks

When the system serves as a Call Center, it will most likely be equipped with more trunks than agent telephones (lines). The reason for having a higher number of trunks is that there are calls in the queue that engage trunk circuits but not ACD agents until they are served. In addition, in a Network Automatic Call Distribution (NACD) application, the overflowed calls continue to occupy trunks without the service of agents at the source node. The trunk-to-agent ratio may change if a service requires a long postcall processing time from an agent. In that case, reduce CCS per agent to reflect the actual agent service times associated with actual calls to the call processor (CP).

Traffic at agent telephones is conservatively assumed to be 33 CCS and 18.3 (= 33 × 100 ÷ 180) calls per agent in the busy hour as a default in examples. For applications with long postcall processing time, default values of 18 CCS and 10 calls per agent are perhaps more appropriate.

Based on the standard system engineering rules, a loop can handle 660 CCS and a superloop can handle 3500 CCS. When an agent is loaded to 33 CCS, a loop can equip 20 agents (= 660 ÷ 33) and a superloop 106 (= 3500 ÷ 33); both numbers are less than their respective number of timeslots (30 for loop, 120 for superloop). Thus, normal network engineering rules do not apply in a Call Center environment, because the "infinite traffic source" assumption in the Erlang model is violated.

Ignore the traffic model here. Instead, use the rule of equipping 30 agents per loop and 120 agents per superloop for a nonblocking connection. The superloop was created to take advantage of the traffic theory that a bigger server group is more efficient than a smaller one. This is no longer true in a nonblocking application, so any superloop can be replaced by four loops without capacity impact. To get the equivalent number of superloops, divide the required number of loops by four.

To calculate loop requirements, treat an agent supervisor telephone like an agent telephone. However, reduce the call intensity of the agent supervisor set for real-time calculations.

The following is the procedure for calculating loop requirements. Let the number of agent telephones be L_1, the number of supervisor telephones be L_2, the number of ACD analog trunks be T_A, and the number of Recorded Announcement (RAN) trunks be T_R:

Number of nonblocking loops for agent telephones, supervisor telephones, and ACD analog trunks = $N_1 = [(L_1 + L_2 + T_A + T_R) \div 30]$

DTI/PRI trunks

At an average of 28 CCS per trunk, a loop of 660 CCS can equip 23 (= 660 ÷ 28) trunks. It is more practical to equip 24 trunks per PRI/DTI loop as a rule, rather than doing traffic calculations.

The equations for trunk loop calculation are as follows. Let T_D be the number of DTI trunks and T_P be the number of PRI trunks:

Number of loops for DTI trunks, $N_{2D} = [T_D \div 24]$

Number of loops for PRI trunks, $N_{2P} = [(T_P + 2) \div 24]$

Number of loops for digital trunks, $N_2 = N_{2D} + N_{2P}$

When a back-up D-channel is not needed, the term $(T_P + 2)$ in the equation for PRI trunks can be replaced by $(T_P + 1)$.

When the number of analog trunks is small (say, 15 or less), it may be included in the N_0 calculation to save loop and slot requirements.

For the international version of PRI, change the number of ports from 24 to 30 in the above calculations. The rest of the engineering procedure is unchanged.

Loops for Avaya CallPilot, Music (MUS), and Music Broadcast applications

Music is provided by broadcasting a music source to a Conference loop. Therefore, a maximum of 30 users can listen to music at one time, which is sufficient for most applications. If not, an additional Conference loop must be provided for each additional set of 30 simultaneous music users.

Music Broadcast requires any Music trunk and an external music source or an Avaya Integrated Recorded Announcer card (NTAG36). Integrated Recorded Announcer has the capability to provide audio input for external music. A Conference loop is not required for Music Broadcast.

Avaya CallPilot ports are interfaced with a loop to provide voice channels for messaging. Each set of 24 ports in the CallPilot interfaces with one loop. The Conference loop connects to one half of the Conference/TDS card. A second Conference loop, if needed, takes another card and card slot, because it cannot be separated from the TDS loop.

The procedure to calculate the number of loops required for applications is as follows:

N_{31} = Music ports ÷ 30

N_{32} = CallPilot ports for CallPilot ÷ 24

Number of loops for applications, $N_3 = N_{31} + N_{32}$

Media Card

The 32-port Media Card provides the DSP gateway function for TDM to LAN/WAN traffic. Calculate N_4, the number of loops required for Media Cards.

Configure no more than four Media Cards on each superloop to eliminate the possibility of blocking because of insufficient timeslots (for example, 4 Media Cards × 32 ports = 128 timeslots).

Physical limits in loops

To calculate N_L, the total number of network loops required in the system, sum the loop requirements previously calculated for agent telephones and ACD analog trunks, digital trunks, applications, and Media Cards:

$N_L = N_1 + N_2 + N_3 + N_4$

Card slots

Calculating the number and assignment of cards and, relatedly, modules can be an iterative procedure, because of specific capacity and usage requirements.

- Card slot usage and requirements on page 202
- Card slot assignment rules on page 203
- Physical slots available to Large System platforms on page 204
- I/O device requirements on page 205
- Card slot calculation algorithm on page 205

Card slot usage and requirements

Table 49: Physical characteristics of Network cards and modules on page 202 summarizes the physical capacities of Large Systemmodules and cards.

Table 49: Physical characteristics of Network cards and modules

Name of card/ module	No. of loops	Card slots	No. of ports/ cards	Comments
NT8D04 Superloop Network card	4	1		Take all 4 loops in two adjacent slots
NT8D17 Conference/TDS	2	1		One (1) card per network module, not separable
NT5D12 DDP card	2	1	2	Needed for PRI/DTI T1s

Name of card/ module	No. of loops	Card slots	No. of ports/ cards	Comments
NT5D97 DDP2 card	2	1	2	Needed for PRI/DTI E1s
SDI (QPC841)		1	4	For MMax (HSL), CDR
MSDL/MISP		1	4	Provides SDI, ESDI and DCHI functions
NT5D21 Core/ Network Module	8	9		Single group system; CC & extra SDI slot in CE
NT8D35 Network Module		8		Multi group system; extra space for single group system
NT4N41	16	9		CPCI Core/Network Card Cage AC/DC for single group and multi group systems

Table 50: System modules on page 203 summarizes the module requirements for various system types.

Table 50: System modules

System	Modules	
	Quantity	Type
Communication Server 1000M SG/ Meridian 1 Option 61C	2	NT5D21 Core/Network Module
Communication Server 1000M SG/ Meridian 1 Option 61C CP PII	2	NT4N41 Core/Network Module
Meridian 1 Option 81C	2	NT5D21 Core/Network Module
Communication Server 1000M MG/ Meridian 1 Option 81C CP PII	2 2	NT4N46 Core/Network Modules or NT4N41 Core/Network Module
This table is for comparison purposes with legacy products. Only the NTN41 Core/Network Module is currently available.		

Card slot assignment rules

The following considerations apply when calculating card slot requirements:

1. The NT5D12 occupies a single network shelf slot. It provides an interface to the 1.5 Mbps external digital line, either directly or through an office repeater, Line

Terminating Unit (LTU), or Channel Service Unit (CSU). The NT5D12 integrates the functionality of two QPC472 DTI/QPC720 PRI cards.

2. A DCHI port is required for PRI. This port can be provided by a DCHI card (with one other port for SDI) or MSDL card (with three additional ports for other functions).
3. A Clock Controller (CC) card is required for PRI or DTI. It has its own dedicated slot on either Core/Network Module of the Meridian 1 Option 61C, or the CE Module of the Meridian 1 Option 81C.
4. A Superloop Network card takes one card slot but, because of address limitations, its adjacent slot cannot be used by any card that requires network loops, such as the Conference/TDS card. The slot next to a Superloop Network card can equip only nonnetwork cards (such as ESDI, MSDL).
5. For the Meridian 1 Option 61C, a Clock Controller is required in slot 9.

Physical slots available to Large System platforms

A Large System network group has 32 loops.

- Each loop has 30 channels.
- Four loops constitute a superloop of 120 channels, with a traffic capacity of 3500 CCS.

Communication Server 1000M SG, Meridian 1 Option 61C

The two NT4N41 Core/Network Modules each have eight card slots for Network and I/O cards. In addition to these slots, there is a fixed slot for a Clock Controller card and an SDI card. It is assumed that one NT8D17 Conference/TDS card is equipped for each Core/Network Module. Without considering applications, the 14 card slots can support 28 traffic loops or 6 superloops.

When H.323 Trunks (Peer H.323 Gateway Trunk) or Session Initiation Protocol Trunks (SIP trunks) are configured on a logic superloop in a Communication Server 1000M SG system, no hardware is required. A superloop of 128 channels can be configured for up to 1024 Virtual Trunks.

Communication Server 1000M MG, Meridian 1 Option 81C

The NT4N41 Core/Network Modules have eight card slots for Network and I/O cards. It is assumed that one NT8D17 Conference/TDS card is equipped for each Core/Network Module. Without considering applications, the 14 card slots can support 28 traffic loops. Additional groups are added for expansion. This configuration consists of a minimum of 2 groups, which contain 28 card slots and can support 56 traffic loops.

The NT8D04 Superloop Network card or NT1P61 Fiber Superloop Network card provides four network loops grouped as one superloop. One superloop can serve up to two NT8D37 IPE Modules. Each superloop provides 120 timeslots and has a traffic capacity of 3500 CCS.

Note that since there are eight card slots in an NT8D35 module, a maximum of three NT8D04 Superloop Network cards, or 12 loops, can be equipped per module if there is also a

Conference/TDS card. This leaves one slot available for the NT5D12 DDP or NT5D97 DDP2 cards, and three slots for I/O cards.

When H.323 trunks or SIP trunks are configured on a logic superloop in a Communication Server 1000M MG system, no hardware is required. A superloop of 128 channels can be configured for up to 1024 Virtual Trunks.

FNF is used to interconnect networks, the maximum size of the system is eight groups with no intergroup junctor blocking.

I/O device requirements

Most advanced features on the system are controlled by auxiliary processors, which communicate with the system CP on routing and other instructions. Since I/O cards compete with network cards for slot space in a network shelf, they are crucial in deciding whether a given system is able to provide all necessary ports and features. Table 51: I/O interface for applications on page 208 summarizes information required to calculate the number of I/O cards needed as an input to the card slot calculation worksheet (see Card slot calculation algorithm on page 205 and Worksheet 19: Physical capacity on page 403).

Certain other applications, such as data, may require interface to I/O ports. However, this section considers only applications that apply in the context of a Call Center.

Once the applications for a given site are known, calculate the required number of I/O ports. Depending on the type of I/O cards provided, determine the number of card slots required.

The system has a maximum of 64 I/O practising MSDL. This constraint may need to be considered for large systems with many application features. For smaller systems, the card slot constraint is a concern, but not the maximum number of I/O addresses.

For Communication Server 1000M SG, MG, and Meridian 1 Option 61C, the ELAN network for the AML/ML interface with a data rate at 10/100 Mbps. Its interface to the system is through an Ethernet connection, and the communication message is an emulation of AML message. The messages from the ELAN network interface continue to interface through CSQI and CSQO, the input/output buffer for AML.

Card slot calculation algorithm

The rules described in this section are summaries of earlier sections. Apply these rules in developing the card slot calculation worksheet (see Worksheet 19: Physical capacity on page 403).

1. Determine Conference/TDS card requirements: one card per Network Module or 14 loops (including virtual loops).
2. Determine MUSic loop card: one Conference/TDS card per music loop.
3. Broadcast RAN does not require CON/Music loop. It broadcasts announcement to waiting calls directly.
4. Each Media Card interfaces with one loop. Each card provides transcoding between 32 TDM channels and 32 DSP channels.
5. Clock Controller slot: put in a zero in this space.
6. Calculate XNET card slots: sum up all XNET loops and divide by 4 to get the card slots required.
7. I/O card slot: the number of slots next to XNET cards that are usable only for I/O cards, regardless of whether needed or not.
8. The sum of the total card slots above should not exceed 16 for Communication Server 1000M SG and Meridian 1 Option 61C CP PIV.

The algorithm described in this section is implemented in the card slot calculation worksheet.

Signaling and data links

Two categories of signaling and data links are discussed in this section:

1. Physical links on page 206
2. Functional links on page 207

Physical links

There are three types of physical links to consider:

- Serial Data Interface (SDI) on page 206
- Multi-purpose Serial Data Link and Multi-purpose ISDN Signaling Processor (MSDL/MISP) on page 207
- Embedded Local Area Network (ELAN) on page 207

Serial Data Interface (SDI)

The SDI is an asynchronous port, providing input access to the system from an OA&M terminal and printing out maintenance messages, traffic reports, and Call Detail Recording (CDR) records to a TTY or tape module. An SDI card has four ports. An MSDL card has four ports for a combination of interfaces.

Multi-purpose Serial Data Link and Multi-purpose ISDN Signaling Processor (MSDL/MISP)

An MSDL card has four ports providing a combination of SDI, ESDI, and DCHI functions. Using MSDL cards, the number of I/O ports in the system can reach 64. If older I/O cards are used, the maximum number per system is 16. The data rate of each port of an MSDL card is dependent on the function it provides. The maximum rate is 64 000 bps for D-channel applications, but lower for other applications.

Embedded Local Area Network (ELAN)

The system can communicate with a Host by Ethernet connection through a Network Interface Card (NIC). AML messages are embedded in the communication protocols, and they continue to interface with the system through CSQI and CSQO queues.

The data rate that can be set at the NIC port is a function of the system CPU type. For CP PIV, the rate can be 10/100/1000MB auto negotiate, half duplex or full duplex.

Functional links

For each of the following functions, the type of link and resulting capacity are given.

High Speed Link (HSL)

The HSL is a an asynchronous link, used for the system CP to communicate with the MAX module via an SDI port. Prior to MAX 8, the HSL bandwidth was 9600. With MAX 8 and later, 19 200 baud is available.

Application Module Link (AML)

AML is a synchronous link between the system and an Application Module (AM) through the ESDI port. The data rate of the link can be one of the following rates: 300, 1.2 KByte, 2.4 KByte, 4.8 KByte, 9.6 KByte, or 19.2 kbps. The standard setup between the system and an AM is the 19.2 kbps link.

OAM

The system uses an SDI port to connect to a computer (TTY) to receive maintenance commands or to print traffic reports, maintenance messages, or CDR records.

ISDN Signaling Link (ISL)

An ISL provides common channel signaling for an ISDN application without PRI trunks. An analog trunk with modems at the originating switch and the terminating switch can be used as an ISL to transmit ISDN messages between these two remote systems. The interface for an ISL is an ESDI port. The maximum data rate for the link is 19.2 kbps.

D-channel

A PRI interface consists of 23 B-channels and one D-channel. The D-channel at 64 kbps rate is used for signaling. A D-channel interfaces with the system through a DCHI card or a DCHI port on an MSDL. A D-channel on a BRI telephone is a 16 kbps link that is multiplexed to make a 64 kbps channel.

Property Management System Interface (PMSI)

The PMSI allows the system to interface directly to a customer-provided PMS through an SDI port. 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 51: I/O interface for applications on page 208 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 51: I/O interface for applications

Application	Type of link/ network interface	Type of port	Sync or async
AML (associated telephone)	AML	ESDI	Sync
Symposium	ELAN	Ethernet	Sync
CallPilot	ELAN	Ethernet	Sync
CDR	RS232 C	SDI	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 MAX	HSL	SDI	Async
Meridian 911	AML	ESDI	sync
Property Management System Interface (PMSI)	PMSI Link	SDI	Async
NACD (PRI)	64 kB D-channel	DCHI	Sync
TTY (OA&M)	RS232 C	SDI	Async
An ESDI card has two ports; an SDI card has two ports; a DCHI card has one DCHI port and one SDI port; an MSDL card has four combination ports.			

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 its line card to a loop through the network to another loop, and on to another line or trunk card attached to the terminating telephone or trunk.

This section discusses the following traffic considerations:

- Loops and superloops on page 209
- Intelligent Peripheral Equipment (IPE) on page 210
- Lines and trunks on page 210
- Groups on page 213
- Service loops and circuits on page 213
- Traffic capacity engineering algorithms on page 216

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 during which a call engages a traffic path or channel
- TRAFFIC – the total occupied time of circuits or channels, generally expressed in CCS or Erlangs (CCS = a circuit occupied 100 seconds; Erlang = a circuit occupied one hour)
- BLOCKING – attempts not accepted by the system due to unavailability of the resource
- OFFERED traffic = CARRIED traffic + BLOCKED traffic
- Traffic load in CCS = Number of calls × AHT ÷ 100 (where AHT = average holding time)
- Network CCS = Total CCS handled by the switching network or CCS offered to the network by stations, trunks, attendants, Digitone receivers, conference circuits, and special features

Loops and superloops

The number of loops needed in the system can be calculated from lines, trunks, and traffic requirements such as average holding time (AHT) and CCS. This section describes the algorithms for these computations, which are incorporated into the traffic worksheet in Worksheet 18: Network loop traffic capacity on page 402.

In the worksheet, the number of lines and trunks are given as inputs. In order to arrive at the number of trunks needed to meet the necessary GoS, the Poisson P.01 table is typically used.

This table can also be used for other circuits requiring P.01 GoS, such as RAN trunks. For the Poisson P.01 table, see Trunk traffic Erlang B with P.01 Grade-of-Service on page 409.

Superloop capacity

Each superloop can carry 3500 CCS of combined station, trunk, attendant console, and Digitone traffic during an average busy season busy hour (ABSBH).

Loop capacity is subject to the Grade-of-Service (GoS) described under Grade-of-Service on page 216.

Intelligent Peripheral Equipment (IPE)

By combining four network loops, the superloop network card (NT8D04) makes 120 timeslots available to IPE cards. Compared with regular network loops, the increased bandwidth and larger pool of timeslots increase network traffic capacity for each 120-timeslot bundle by 25% (at a P.01 GoS). The recommended traffic capacity for an IPE superloop is 3500 CCS, which meets all GoS requirements for network blocking. For nonblocking applications, a superloop can be equipped up to 120 lines or trunks, and each circuit can carry up to 36 CCS.

Lines and trunks

The relationship between lines and trunks is relevant for calculating loop requirements. Figures Figure 56: Traffic calls TDM only on page 211 and Figure 57: Communication Server 1000M system call types on page 212 show how traffic parcels on a loop can be broken up.

Figure 56: Traffic calls TDM only on page 211 represents traffic in a Meridian 1 TDM-based environment. For additional information, see Peripheral Equipment on page 65.

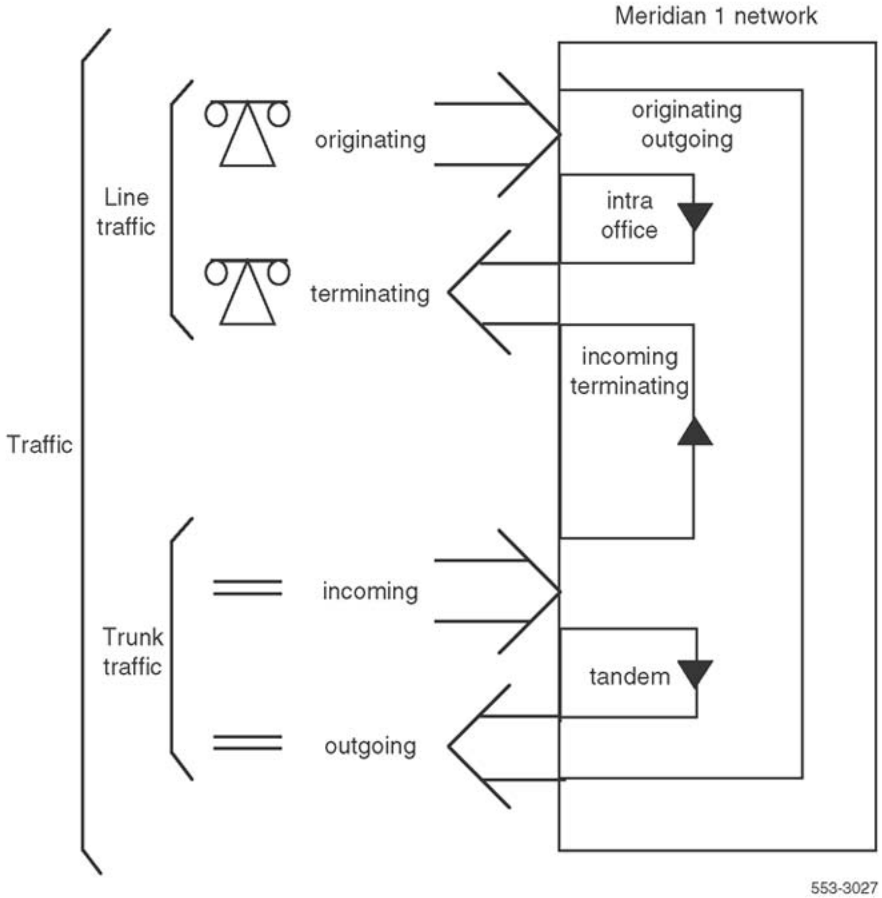

Figure 56: Traffic calls TDM only

Voice over IP traffic

In the context of Voice over IP (VoIP) application, the lines include IP Phones (UNIStim, SIP Line, and SIP Dect) 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 57: Communication Server 1000M system call types on page 212 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, TDM to TDM connections require no DSP, and neither do IP to IP or IP to VT.

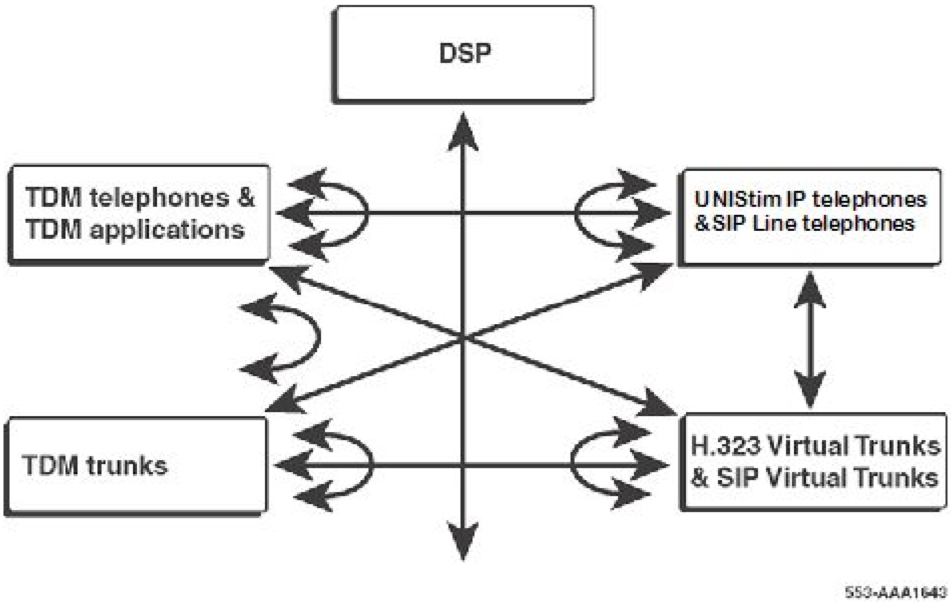

Figure 57: Communication Server 1000M system call types

Table 52: Connection type resources required on page 212 lists the resources required for each type of connection.

Table 52: 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, VT to UNIStim IP	VT
TDM to TDM telephone or trunk calls	no DSP, no VT
TDM to SIP Line, SIP Line to TDM	DSP, VT
SIP Line to SIP Line	VT
SIP Line to VT, VT to SIP Line	VT
SIP Line to UNIStim IP, UNIStim IP to SIP Line	VT

See Resource calculations on page 247 for the algorithms to calculate the required resources.

Groups

A network group is comprised of two network modules of 16 loops each, for a total of 32 loops. The maximum size of a system expanded with Fiber Network Fabric (FNF) is 8 groups or 256 loops.

There are two types of loops:

- Terminal loops provide channels for general traffic.
- Service loops provide tones and service functions.

The number of groups in a system is determined by the number of terminal loops and service loops required (see Card slot usage and requirements on page 202).

Service loops and circuits

Service circuits are required in call processing to provide specific functions to satisfy the requirements of a given application. They are system resources. Service circuits consume system resources, such as physical space, real time, memory, and so on.

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

- TDS on page 213
- Conference on page 214
- Broadcast circuits on page 214
- DTR on page 215

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.

In other words, the maximum number of simultaneous users of tone circuits is 30, whether it be 30 of one tone or a combination of many different types of tones. One TDS loop is normally recommended for each Network Module or half network group of 14 traffic loops. Additional TDS loops may be added if needed, but this is rare.

Conference

The Conference (CON) loop is a part of the dual loop NT8D17 Conference/TDS card. It provides circuits for 3-way or 6-way conferences. It can also broadcast music from a source to a maximum of 30 users simultaneously. In addition, a CON loop also provides temporary hold for a variety of features, chief among them, the End to End Signaling. One CON loop is normally recommended for each half network group or 14 traffic loops.

Music

Music can be provided through conferencing a caller to a MUS source. Therefore, a CON loop is required for the Music on Hold feature. Each set of 30 simultaneous music users will require a CON loop, thus a Conference/TDS card, since these two service loops are not separable.

Use the following formula to calculate MUS traffic:

MUS CCS = Number of ACD calls using MUS × MUS HT ÷ 100

A segment of music typically runs from 40 seconds to 60 seconds. If the average for a specific application is not known, use a default of 60 seconds. After CCS is obtained, estimate the MUS port requirement from a Poisson P.01 table or a delay table (such as DTR table) matching the holding time of a MUS segment (see Reference tables on page 409).

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.

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 Communication Server 1000Msystem to be used as the music source on demand to all parties on hold. With Network Music, the Communication Server 1000Msystems 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. DTRs are system-wide resources and should be distributed evenly over all network loops.

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

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 the number of loops and system size.

Alternatively, when the loop 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:

1. Dial tone delay is not greater than 3 seconds for more than 1.5% of call originations.
2. The probability of network blocking is 0.01 or less on line-to-line, line-to-trunk, or trunk-to-line connections.
3. Blocking for ringing circuits is 0.001 or less.
4. postdialing delay is less than 1.5 seconds on all calls.

Any connection in the system involves two loops, one originating and one terminating. In an intergroup connection of a multi-group system, it also involves an intergroup junctor, which can also incur blocking. Each stage of connection is engineered to meet 0.0033 GoS. Therefore, overall network blocking in the system is less than 0.01, regardless of whether the call is a line or trunk call, or an intra- or intergroup call.

There is no intergroup blocking for the eight-group network with fiber junctors.

Traffic models

Table 53: Traffic models on page 217 summarizes the traffic models that are used in various subsystem engineering procedures.

Table 53: 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. For a description of the TFS004 call capacity report and for information about interpreting TFS004 output, see *Avaya Traffic Measurement Formats and Outputs References, (NN43001-750)*.

If a new switch is being configured, equivalent basic calls must be calculated in order to estimate the processor loading of a proposed configuration.

Equivalent Basic Calls

An Equivalent Basic Call (EBC) is a measure of the real time required to process a basic call. A basic call is defined as a simple, unfeatured call between two 2500-type telephones on the same switch using a four-digit dialing plan. The terminating telephone is allowed to ring once, then is answered, waits approximately two seconds, and hangs up. The originating telephone then hangs up as well.

When the capacity of a switch is stated in EBC, it is independent of such variables as configuration, feature mix, and usage patterns. It still varies from release to release, and between processors. However, since it is independent of other factors, it is a good way to compare the relative call processing capability of different machines running the same software release.

Table 54: Real-time capacity (EBC) by system on page 218 gives the rated capacities of the Call Server processors in Large Systems operating Communication Server 1000 software.

Table 54: Real-time capacity (EBC) by system

System	Capacity
Communication Server 1000M/Meridian 1 PBX with CP PIV	741 000

Feature impact

Every feature that is applied to a call increases the CP real time consumed by that call. These impacts can be measured and added incrementally to the cost of a basic call to determine the cost of a featured call. This is the basis of the algorithm used by EC to determine the rated capacity of a proposed switch configuration.

The incremental impact of a feature, expressed in EBC, is called the real-time factor for that feature. Real-time factors are computed by measuring the incremental real time for the feature in milliseconds, and dividing by the call service time of a basic call.

Each call is modeled as a basic call plus feature increments. For example, an incoming call from a DID trunk terminating on a digital telephone with incoming CDR is modeled as a basic call plus a real-time increment for incoming DID plus an increment for digital telephones plus an increment for incoming CDR.

A second factor is required to determine the overall impact of a feature on a switch. This is the penetration factor. The penetration factor is simply the proportion of calls in the system that invoke the feature.

The real-time impact, in EBC, of a feature on the system is computed as follows:

(Calls) × (penetration factor) × (real-time factor)

The sum of the impacts of all features, plus the number of calls, is the real-time load on the system, in EBC.

For penetration and real-time factors and for the detailed EBC calculations, see System calls on page 251 and Real-time calculations on page 255.

Call Server real-time calculations

The system EBC divided by the processor's rated capacity (see Table 54: Real-time capacity (EBC) by system on page 218) yields the fraction for processor utilization. This determines whether the proposed system will 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 will 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.

I/O impact

There are two types of I/O interface allowed at the system: the synchronous data link and asynchronous data link. ESDI and DCHI cards provide interface to synchronous links, and an SDI card provides interface to asynchronous links. The MISP/MSDL card can provide both.

At the I/O interface, the system CP processes an interrupt from SDI port per character while processing an ESDI/DCHI interrupt per message (multiple characters). As a result, the average real-time overhead is significantly higher in processing messages from an SDI port than from an ESDI port. MSDL, however, provides a ring buffer.

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 311.

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 will do, 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, if applicable, see Table 60: Resource calculation parameters on page 248.

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 may be necessary to separate a single type into low-usage and high-usage categories. For example, a typical office environment with analog telephones may have a small call center with agents on analog 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
- Attendant consoles
- DECT handsets
- IP Phone 200x, IP phone 11xxE
- Avaya 2033 IP Conference Phone
- Avaya 2050 IP Softphone
- WLAN Handset 2210, 2211 and 2212
- 802.11 Wireless LAN terminals

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 may 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 of which provides 24 trunks under the North American standard
- PRI: number given in terms of links, each of which provides 23B+D under the North American standard
- European varieties of PRI: VNS, DASS, DPNSS, QSIG, ETSI PRI DID

H.323 Virtual Trunk

An IP Peer H.323 Virtual Trunk identified with a trunk route which is not associated with a physical hardware card.

SIP Virtual Trunk

A Session Initiation Protocol (SIP) Virtual Trunk identified with a trunk route which 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 AHTRAN is 30 seconds.

Music

The default value for AHTMUSIC 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
- Unified Communication Manager (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 maximum call rate, see Table 68: Signaling Server algorithm constants on page 270.

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 232.

Table 55: Elements in Signaling Server on page 223 describes the function and engineering requirements of each element.

Table 55: 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.

Element	Function and engineering requirements
	• 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 H.323 calls (see Maximum number of SIP and H.323 Virtual Trunks on page 231). 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 Communication Server 1000M 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, COTS2 and Common Server, 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 231). • 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 for CP DC, COTS2 and Common Server, 1800 for CP PM and COTS1. The maximum SIPL users for a Communication Server 1000M Call Server is 2500 single group, and 7500 multiple group. • 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

Element	Function and engineering requirements
	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 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.

Element	Function and engineering requirements
• 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.
• 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 Media Gateway 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.

Element	Function and engineering requirements
	• 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.

Table 56: 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

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
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
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	N/A	N/A	800 IBM 700 DELL	800	240
MAS cph	N/A	N/A	6000	6000	2000
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

Table 57: 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

	CP PM	COTS1 (HP DL320-G4, IBM x306m)	COTS2 (Dell R300, IBM x3350)	Common Server (HP DL360-G7)	CP DC
MSC IPTone sessions	800	1000	1000	1000	1000
MSC IPAttn sessions	256	256	256	256	256
MSC total sessions **	800	1000	1200	1200	1000
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

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 58: Maximum number of Virtual Trunks, per Signaling Server on page 231 gives examples of the maximum number of Virtual Trunks supported for different configurations.

Table 58: 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 coreside with other applications (TPS, H.323/SIP Gateways, Element Manager). For more information on Co-resident limits, see Table 68: Signaling Server algorithm constants on page 270. 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 183 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 Media Cards per node is 30. When more than 30 Media Cards are needed on a single Communication Server 1000M Large System, use multiple nodes. The maximum number of Signaling Servers and Media Cards combined within a node is 35.

Communication Server 1000M capacities

Since IP telephony consumes more 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 59: Communication Server 1000M Large System traffic capacities summary on page 233 summarizes the capacities of Communication Server 1000M Large 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 59: Communication Server 1000M Large System traffic capacities summary

Call server	Platform name	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
CP PIV	Communication Server 1000M SG	3000	15 000	2x(SipN +Sip3) <= 3000	(UNIStim + 2x(SipN +Sip3) + SipDect + TDM) <= 3000
CP PIV	Communication Server 1000M MG	16 000	15 000	2x(SipN +Sip3) <= 15 000	(UNIStim + 2x(SipN +Sip3) + SipDect + TDM) <= 15 000

Call server	Platform name	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
Values in each column reflect the total telephones for a configuration. These are absolute limits for pure TDM and pure IP. For mixed TDM and IP, values are for typical configurations. Applications and calling patterns impact call server capacity. EC and publications are used to calculate practical values preconfiguration. Values beyond these limits must be engineered. Requires using Signaling Servers for TPS. IP telephones with access to PSTN and the mixed configurations assume 8-15% digital trunking to PSTN and no applications.					

Zone/IP Telephony node engineering

Zone/IP Telephony Node engineering is a network function which controls network response to traffic demands and other stimuli, such as network failures. This engineering encompasses:

- traffic management through control of routing functions
- capacity managements through control of network design
- traffic measurement and modeling
- network modeling (example: load balancing, scalability, reliability, redundancy)

Zone node engineering

A network zone is a logical grouping of CS 1000M systems with IP Peer H.323 Gateways, IP Line, IP Trunk 4.0 (and later), and/or third-party gateways or endpoints.

Network zones can have geographical significance; for instance, a company could configure one network zone for its east coast offices, and one network zone for its west coast offices.

Routing (SIP/H.323) Zones

In a SIP/H.323 network, each NRS controls one SIP/H.323 zone. Each zone can consist of many SIP/H.323 endpoints. If a call terminates beyond the call originator's own zone, the Redirect Server or H.323 Gatekeeper of the called party's zone provides the endpoint information to set up the connection.

Network Bandwidth Management

To optimize IP Line traffic bandwidth use between different locations, the IP Line network is divided into zones, representing different topographical areas of the network. All IP Phones and IP Line ports are assigned a zone number indicating the zone to which they belong. When a call is made, the codecs that are used vary depending on the zone(s) in which the caller and receiver are located.

By default, when a zone is created in LD 117:

- codecs are selected to optimize voice quality (BQ - Best Quality) for connections between units in the same zone
- codecs are selected to optimize voice quality (BB - Best Bandwidth) for connections between units in different zones

Each zone can be configured to:

- optimize either voice quality (BQ) or bandwidth usage (BB) for calls between users in that zone
- optimize either voice quality or bandwidth usage within a zone and all traffic going out of a zone

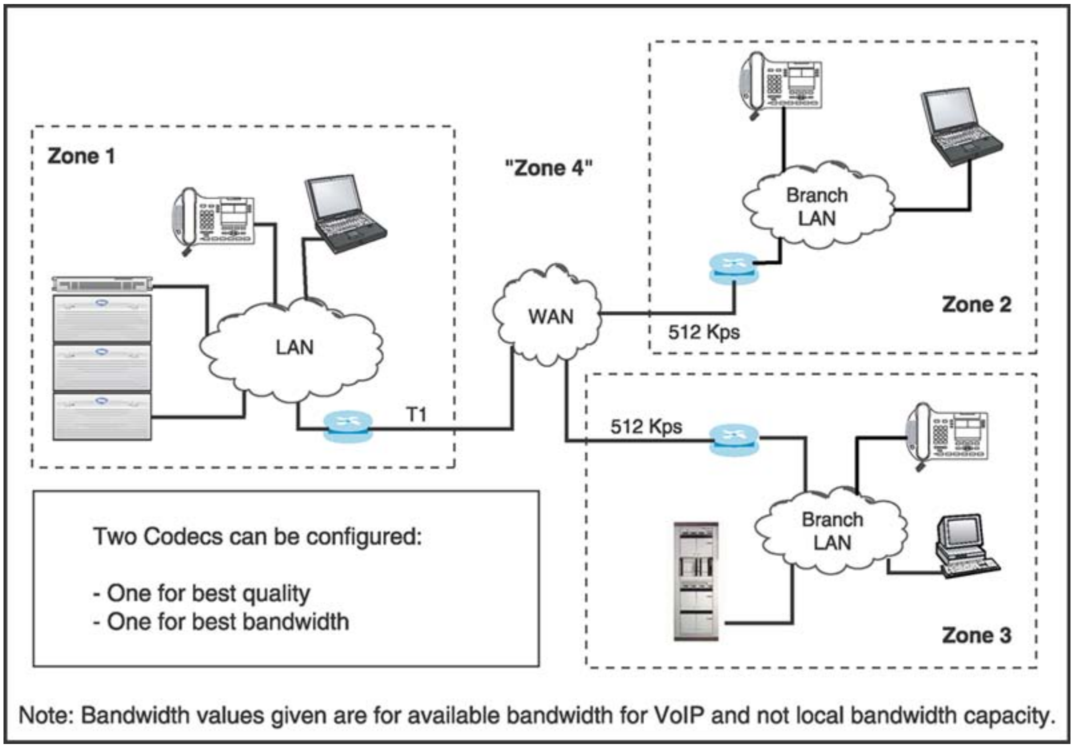

Figure 58: Bandwidth management example

CS 1000 provides support for bandwidth management on a network-wide basis so that voice quality can be managed between multiple Call Servers using IP Peer Networking in certain scenarios.

The Network Bandwidth Management feature allows bandwidth zones to be configured on a network basis so that codec selection and bandwidth allocation software can identify whether IP Phones or gateways are physically co-located (in the same bandwidth zone) even though they are controlled by different Call Servers.

An IP Peer network is divided into different bandwidth management zones. Each IP Phone, Virtual Trunk, or Voice Gateway DSP channel is assigned to a bandwidth management zone. All IP Phones, Virtual Trunks, or Voice Gateway DSP channels in a bandwidth management zone:

- share the same IP bandwidth management policies
- are geographically near each other
- are all in the same time zone
- are all in the same PSTN dialing plan

A bandwidth management zone is assigned to each Virtual Trunk and Voice Gateway DSP Channel in LD 14. It is assigned in the same way as the ZONE for an IP Phone in LD 11. This zone enables the trunk to send a setup message, with a codec list selected according to the Best Bandwidth (BB) or Best Quality (BQ) criteria for that zone.

For dialing plan purposes, all telephones in the same zone can be treated identically. Each IP Phone is assigned to a zone during configuration and different zone numbers are assigned to different MG 1000 systems.

Customer zones

It is possible to divide a customer within a system into different zones; however, it is more common to assign one zone to one system and one customer.

Private/shared zones

The IP phones configured in shared zones use DSP resources configured in shared zones. If all of the shared zones' gateway channels are used, the caller receives an overflow tone and the call is blocked. The order of channel selection for the gateway channels is:

1. a channel from same zone as IP Phone is configured
2. any available channel from the shared zones' channels

New zone types introduced by IPL 3.0. DSP channels and configured in a private zone are only used by IP Phones that have also been configured for that private zone.

If more DSP resources are required by these IP phones than what is currently available in the zone, DSPs from other zones are used. However, IP Phones configured in shared zones cannot use private zone channels. The order of selection for the gateway channels is:

1. a channel from same private zone as IP Phone is configured
2. any available channel from the pool of shared zones' channels

Zones and branch office locations

Bandwidth zones can be configured on a network basis if Avaya MG 1000B IP Deskphones are controlled by a main office Call Server. In this configuration, all TDM devices (such as digital and analog 500/2500-type telephones and trunks to the local PSTN) are under the control of the Avaya MG 1000B MGC.

In this case, calls from IP Phones to these TDM devices do not use any LAN/WAN (Interzone) bandwidth for media and should, therefore, use the Intrazone algorithms for bandwidth allocation and codec selection policy. Network Bandwidth Management provides a mechanism to identify this configuration and adjust the algorithms accordingly. Once all bandwidth is used, any additional calls are blocked.

To implement this feature, the Virtual Private Network Identifier (VPNI) prompt exists in LD 15. This enables the bandwidth management feature and expands the number of bandwidth zones beyond the current maximum of 256. When VPNI is set to its default value of 0, Network Bandwidth Management is disabled.

Relationship Between Zones and subnets

IP Phones and Media Cards gateway ports are assigned to zones based on the bandwidth management requirements of the particular installation. Devices in different subnets must traverse a router to communicate, and can reside on different ends of a WAN facility. When IP Phones and gateway ports are in different subnets, the network facilities between them must be examined to determine if placing the separated devices in different zones is warranted.

It is not necessary to always assign different zones. For instance, there can be different subnets within a LAN interconnected by router(s) with sufficient bandwidth. The IP Phones and gateway channels spread across them could all reside in a single zone. However, if there is a WAN facility with limited bandwidth between two subnets, the devices on the opposite ends should be placed in different zones so the bandwidth across the WAN can be managed.

For remote users such as telecommuters, bandwidth management is not normally a consideration because only one IP Phone is present at the remote location. It can be convenient to allocate zones for users with similar connection speeds. In that case, set both the interzone and intrazone codec to Best Bandwidth (BB).

IP Telephony Node

An IP Telephony node is defined as a collection of Signaling Servers and or Media Cards. Media Cards only provide DSP resources, they do not support the IP Line application. Each node in a network with one or more call servers has a unique Node ID. This Node ID has an integer value. A node has only one Leader Signaling Server. All the other Signaling Servers and Media Cards are defined as Followers.

A node can have a practical maximum of 5 Signaling Servers, and a practical maximum of 30 Media Cards for an actual maximum of 35.

The TPS uses the Node ID and Node IP address for IP set registration. SIP/H.323 Gateways use Node IP address, as well as NRS Manager in case of Gateways being static endpoints.

The Node ID of SIP/H.323 Gateway has to be entered in the Route Data Block in Overlay 16 on Call Server side. The NRS does not use the Node ID or the Node IP address. A call server supports multiple nodes.

TPS

One node can have one Signaling Server that acts as a Leader or TPS master, and within the same node there can be multiple Signaling Servers acting as Followers. The IP Phones are distributed between the Signaling Servers (load-sharing). The Node number is programmed into the IP set.

Media Card is a term used to encompass the Media Card 32-port secure line card, Media Card 32-port line card, and Media Card 8-port line card. These cards plug into an Intelligent Peripheral Equipment (IPE) shelf in the Meridian 1 and CS 1000M systems. They also plug into Avaya MG 1000E for the CS 1000E systems.

All IP Phones register with the Signaling Server. The Media Cards only provide access to the voice gateway (DSP resource), and do not provide LTPS service. When present, the Signaling Server is the node leader and acts as a Master for the node.

The H.323 Gateway runs on the Leader Signaling Server. The maximum capacity for a standalone H.323 Gateway Signaling Server is 1200 H.323 virtual trunks.

In a case where the number of H.323 virtual trunks is greater than the Co-resident limit, the H.323 Gateway cannot coreside with other Signaling Server applications. In this case, you must configure a standalone Signaling Server in a new node. For the H.323 virtual trunk Co-resident limit, see Table 68: Signaling Server algorithm constants on page 270.

The SIP Gateway runs on the Leader Signaling Server. The maximum capacity for a standalone SIP Gateway Signaling Server is 1800 SIP virtual trunks.

In a case where the number of SIP virtual trunks is greater than the Co-resident limit, the SIP Gateway cannot coreside with other Signaling Server applications. In this case, you must configure a standalone Signaling Server in a new node. For the SIP virtual trunk Co-resident limit, see Table 68: Signaling Server algorithm constants on page 270.

In a Communication Server 1000 system H.323 or SIP Gateways cannot share among multiple customers. If there are multiple customers, you must configure a H.323 or SIP Gateway for each customer.

A Signaling Server can support multiple routes from one customer. You configure the routes to use the same node ID and the same D channel.

You must configure the PD/CL/RL on a Leader Signaling Server with node ID. Also, the Network Routing Service (NRS) must run on a Leader Signaling Server with node ID.

The Signaling Server applications can run Co-resident on one Signaling Server if the individual application does not exceed the Co-resident limit. If a Signaling Server application exceeds the Co-resident limit, you must deploy that application on a standalone Signaling Server. For more information about the Co-resident limit, see Table 68: Signaling Server algorithm constants on page 270.

Node Redundancy

Signaling Server redundancy ensures that telephony services can withstand single hardware and network failures. It also provides a load-sharing basis for the Terminal Proxy Server (TPS) and an alternate route for the SIP and H.323 Gateway software. When planning survivability strategies for the Signaling Server, one or more additional Signaling Servers should be included in the plan.

The redundancy of TPS is N+M. Therefore, extra Signaling Servers can be provided to cover TPS functions from N other servers. With a redundant Load Sharing Signaling Server:

- One or more Signaling Servers can be configured in a normal configuration.
- The redundant Signaling Servers must be configured in the same TPS node as the Signaling Servers they are protecting.
- If any of the Signaling Servers fails, IP Phones that were registered to the failed Signaling server register to the remaining Signaling servers in the same node.

The redundancy of Media Cards is N+M. The limit of 30 Media Cards per node does not impact redundancy. If the DSP on the Media Card are configured in a shared zone, then they are accessible by all applications.

In a multi-customer situation, the individual DSP channels on the card can be assigned to any customer and they cannot be shared between customer. Redundancy must be calculated on an individual customer basis.

The redundancy mode of the H.323 or SIP Gateway is either 1:1 or 1 + M. In the 1:1 configuration one Gateway is in an active state and the other is in a standby state. If the active Gateway fails, the standby Gateway becomes active. In the 1 + M configuration, you configure M additional Signaling Servers each as Leaders on a new node. 1 + M configuration requires careful ESN provisioning to support load balancing and redundancy. If a Signaling Server fails in a 1 + M configuration, redundant routes are immediately available to carry trunk traffic. The 1 + M configuration requires additional virtual trunk license costs.

The PD/RL/CL application Signaling Server does not support redundancy. A backup and restore function is available to preserve customer data. If a PD/RL/CL application Signaling Server fails, it does not impact call processing.

For information about redundancy on NRS, see Table 55: Elements in Signaling Server on page 223.

Multi-Node configuration

In the event that an application goes beyond its capacity, a separate node is required. There is no known limit to the number of nodes supported by a single call server. The practical limit based on what could be configured and the number of unique nodes required is relatively small, so there is no scaling limit imposed by the number of nodes on a single call server. In the event that a customer needs more than 30 Media Cards, a separate node must be configured.

For Signaling Server Co-resident application limits, see Table 68: Signaling Server algorithm constants on page 270.. In a situation where these numbers are exceeded, a separate node is required to run H.323/SIP Gateways, PD/RL/CL, or NRS applications.

If a customer requires more than 1800 SIP trunks or 1200 H.323 virtual trunks, a separate node is required in order to run H.323/SIP Gateway applications. This is due to a restriction that a node can have only one Leader Signaling Server.

In multi-customer configuration, it is required to create a separate node for virtual trunks Gateways (SIP/H.323). Multi-customer configurations require separate DSP channels; however, these channels can be on the same Media Card.

Preferred performance

In the event that a performance criteria must be met, a separate node creation may be required to accommodate those requirements.

Example 1

The following example explains a possible configuration between two Meridian 1/CS 1000M switches to achieve both resiliency in the IP network, and load balancing.

Meridian 1/CS 1000M switch A has two IP Trunk nodes, A1 and A2, for the destination NPA 613. A Route List Block (RLB) is created in order to have two route entries (one for each IP Trunk node).

If the trunks of node A1 are all in use, or node A1 is down, call traffic is routed to node A2. This provides resiliency by preventing failure of a single IP Trunk node (for example, DCH failure or Leader subnet fails) from completely eliminating VoIP service for a Meridian 1/CS 1000M system.

It is desirable to distribute calls to multiple nodes at a remote destination Meridian 1/CS 1000M. The configuration of multiple dialing plan entries at the local IP Trunk node allows routing based on the dialed digits.

For example, Meridian 1/CS 1000M switch B node B1 has two entries for NPA 408 and 4085, which point to nodes A1 and A2 of Meridian 1/CS 1000M switch A, respectively. Calls from B1 with dialed digits 408-5xx-xxxx are routed to the IP Trunk node A1 while all other 408-xxx-xxxx calls are routed to IP Trunk node A2.

Example 2

In order to speed up IP sets registration, a separate node may be created with a SS running TPS application, that would handle high priority phones, and speed up their registration.

Branch office - node relationship

A node is not split between branches.

Limits of a single node

The number of Media Cards and Signaling Servers combined in a node is limited to 35, without exceeding the limitations of each element type. The maximum number of Media Cards for each node is 30 while the maximum number of Signaling Servers for each node is 35.

Statistics, error, or log files relative to a node

All statistics, error and log files are for a single node, and are available through Element Manager. There are no summary reports or statistics for multiple nodes.

Node management

Node management is performed through Element Manager.

IP address requirements

Each card within a node has two IP addresses:

1. TLAN network interface and for the Meridian 1
2. CS 1000M ELAN network interface

Each node has one Node IP address on the TLAN subnet that is dynamically assigned to the connection server on the node Master. The Internet Telephone uses the Node IP address during the registration process. All CS 1000 ELAN network interface IP addresses must be on the same subnet as the system Call Server ELAN network interface IP address.

For more information about Zone/IP Telephony node engineering, refer to the following documents:

- *Avaya Converging the Data Network with VoIP Fundamentals, (NN43001-260)*
- *Avaya Branch Office Installation and Commissioning, (NN43001-314)*
- *Avaya IP Trunk Fundamentals, (NN43001-563)*
- *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*
- *Avaya SIP Line Fundamentals, NN43001-508*

Chapter 13: Large System Redundancy Aspects

The chapter covers various aspects that should be considered while building a large system. These considerations should be met to provide redundancy to the switch.

Core card placement

The Large System has two call processor (core) cards placed in the two core/net shelves, one acting as an active core and other in the standby mode listening to the active core. The cores must always be placed in two different columns. This ensures that when one column breaks, the other core takes control of the entire system.

The following figure shows an ideal configuration for core card placement.

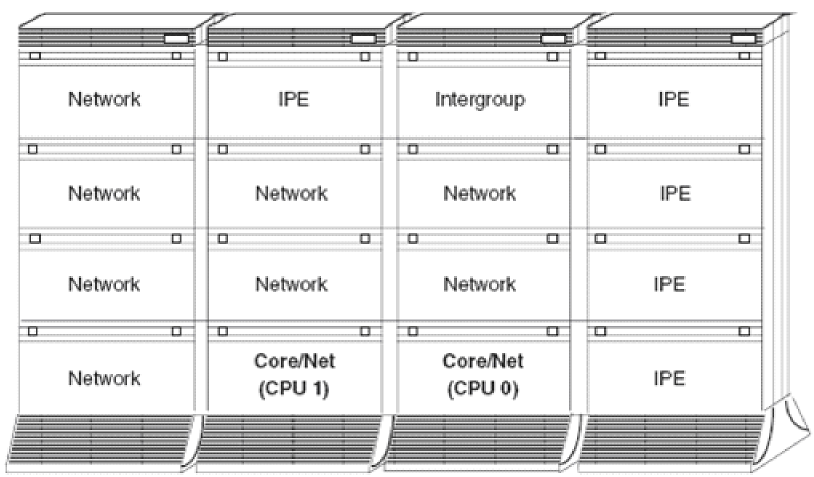

Figure 59: Ideal Configuration for Core Card Placement

FIJI card placement

FNF architecture has two SONET rings, and each FIJI in the system is recognized by its group and side. Both the rings carry half the traffic of the entire switch. In a redundant switch, both the rings should be HALF-HALF. When one of the rings break, they go to FULL-NONE or NONE-FULL state. When both the rings break, they go into SURVIVAL mode. There must not be two FIJI cards in one column which belong to the same group. For example: Column 0 must not contain two FIJIs which belong to one group, say G3. Instead of having G3S0 and G3S1 FIJIs in the same column 1, split them in column 1 and 2. This applies to all FIJIs and all columns.

The following figure shows a wrong way of placing the FIJIs. In this case, when the column breaks down, both the FIJIs are down, this in turn causes both the rings to be in survival state thereby affecting the entire traffic.

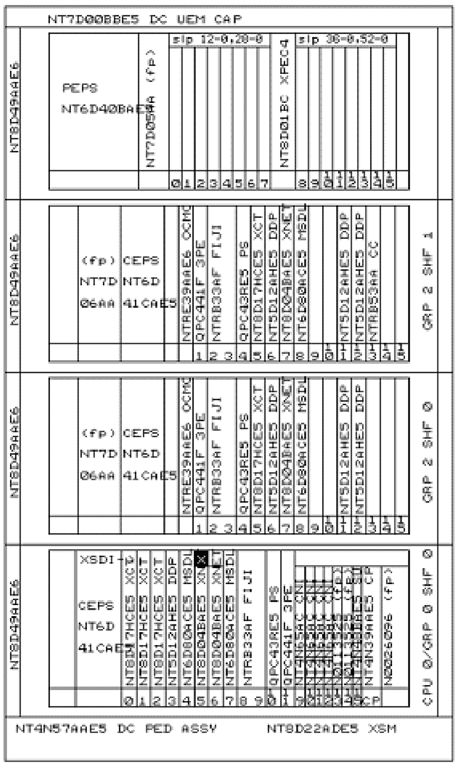

Figure 60: Wrong placement of FIJIs

In a redundant switch, G2S0 and G2S1 must not be in the same column.

Clock controller placement

There are two clock controller cards in the large system, one of them is an active card and the other is in standby mode. The active clock controller tracks to the external reference and supplies clock to the entire switch. This clock controller also provides clock to the inactive or standby clock. Both the clock controller cards should be placed in two different columns to provide a completely redundant switch. In addition to this, one of the each clock controllers must be placed on both the sides of the FIJI ring, one corresponding to each ring.

PRI card placement

The clock controller packs in the large system will be tracking to two PRI cards which are connected to external references. One of the PRI card is designated as primary reference and the other one as designated as secondary reference. The PRI cards, which the clock controllers are tracking must be placed in two different columns.

Clock controller hardware vintages

The vintages of the clock controller cards should match to support redundancy. Unmatched clock controller card vintages can cause problems during switchover if there is a different patch or firmware on the card. This difference can cause a disturbance in the traffic and performance of the switch.

Chapter 14: Resource calculations

Contents

This chapter contains the following topics:

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Resource calculation equations on page 249

System calls on page 251

Real-time calculations on page 255

DSP/Media Card calculations on page 261

Virtual Trunk calculations on page 267

Signaling Server algorithm on page 269

Reducing imbalances (second round of algorithm calculations) on page 292

Illustrative engineering example on page 294

Introduction

This chapter describes the algorithms implemented by the EC tool in order to calculate the resources required by the system.

In many cases, the calculations require user inputs that are the result of preengineering performed in accordance with the capacities and guidelines described in System capacities on page 193 and Application engineering on page 311.

When a proposed new system will be equipped with more ports than the initial configuration will actually use, 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 60: Resource calculation parameters on page 248 describes the parameters you use in resource calculations.

Table 60: Resource calculation parameters

Parameter	Description	Default value
$Telephone_{CCS}$	Common Channel Signaling (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 Avaya CallPilot port	26 CCS
$CP2_{CCS}$	CCS for each network Avaya 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 Avaya 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 61: Resource calculation equations on page 249 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 61: 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. Intraoffice calls (C_{SS}) on page 252 (telephone-to-telephone)
2. Tandem calls (C_{TT}) on page 253 (trunk-to-trunk)
3. Originating/outgoing calls (C_{ST}) on page 253 (telephone-to-trunk)
4. Terminating/incoming calls (C_{TS}) on page 254 (trunk-to-telephone)

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)

      $C_{T2HS} = CTT \times V^2 \times V_H \times V_S \times 2 \times 2$ (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 258
- Feature and applications EBCs on page 258
- Call Server utilization on page 259
- CPU real-time conversion for upgrades on page 259

The real time required to process a basic 2500-type telephone to 2500-type telephone call is an Equivalent Basic Call (EBC), the unit used to measure other, more complicated feature calls. Every feature call can be converted to EBCs by using its real-time factor (RTF).

RTF = (Real time of a feature call in ms ÷ Real time of a basic call) - 1

There are a total of 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 62: Real-time factors on page 256 provides the real-time factors.

Table 62: Real-time factors

Type of call	Real-time factor	Avaya CS 1000M
Intraoffice calls:		
UNIStim IP telephone to UNIStim IP telephone	(f_1)	0.13
UNIStim IP telephone to TDM telephone	(f_2)	1.08
TDM telephone to TDM telephone	(f_3)	0.03
Tandem calls:		
Virtual Trunk to TDM trunk	(f_4)	1.83
TDM trunk to TDM trunk	(f_5)	2.07
H.323 Virtual Trunk to SIP Virtual Trunk	(f_6)	1.60
Originating/outgoing calls:		
UNIStim IP telephone to Virtual Trunk	(f_7)	1.33
UNIStim IP telephone to TDM trunk	(f_8)	2.00
TDM telephone to Virtual Trunk	(f_9)	1.75
TDM telephone to TDM trunk	(f_{10})	2.30
Terminating/incoming calls:		
Virtual Trunk to TDM telephone	(f_{11})	1.84
Virtual Trunk to IP telephone	(f_{12})	1.87
TDM trunk to UNIStim IP telephone	(f_{13})	2.13
TDM trunk to TDM telephone	(f^{14})	1.40
SIPL to SIP telephone	(f^{15})	2.90
SIPL to UNIStim IP telephone	(f^{16})	1.52
SIPL to TDM telephone	(f^{17})	1.48
SIPL to Virtual Trunk	(f^{18})	3.12
SIPL to TDM Trunk	(f^{19})	3.53
Virtual Trunk to SIP telephone	(f^{20})	2.30
TDM Trunk to SIPL	(f^{21})	2.27
Application/feature calls:		
ACD agent without Symposium	(f_{ACD})	0.13
Symposium	(f_{SYM})	5.70
CallPilot	(f_{CP})	1.70
Avaya Integrated Conference Bridge	(f_{MICB})	1.59

Type of call	Real-time factor	Avaya CS 1000M
Avaya Integrated Recorded Announcer	(f_{MIRAN})	0.63
Avaya Integrated Call Assistant	(f_{MICA})	0.57
Avaya Hospitality Integrated Voice Service	(f_{MIVS})	0.57
Avaya Integrated Call Director	(f_{MIPCD})	0.63
BRI ports	(f_{BRI})	0.12
MDECT telephone	(f_{DECT})	4.25
Intraoffice CDR	(f_{ICDR})	0.44
Incoming CDR	(f_{CCDR})	0.32
Outgoing CDR	(f_{OCDR})	0.32
Tandem CDR	(f_{TAN})	0.44
CPND factor	(f_{CPND})	0.20
Converged Desktop factor	(f_{DTP})	2.33
Error term – minor feature overhead	(f_{OVRH})	0.25
Microsoft Converged Office factor	(f_{mo})	2.33
IP Security factor	(f_{IPSEC})	0.33
MC3100 factor	(f_{MC3100})	4.78
MobileX factor	($f_{mobileX}$)	0.67
Extend Local Calls factor	(f_{ELC})	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 252 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 63: Feature and applications EBC on page 258 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 63: 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}
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}

EBC feature or application	Calculation
Hospitality Integrated Voice Services	Number of Hospitality Integrated Voice Services ports × CCS × 100 ÷ AHT_{MIVS} × f_{MIVS}
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}
ITG Trunk (only for 11C,61C,81C)	[(ITG ports × 28) × 100 ÷ 150] × f_{ITG}
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 telephones and extensions)
MobileX	(C_{ss} + C_{TT} + C_{ST} + C_{TS}) × MobileX_P × $f_{MobileX}$ where MobileX_P = MobileX_users / (total number of telephones and extensions)
Extend Local Calls (EBC)	(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 + ITG_EBC + CD_EBC + MO_EBC + IPSEC_EBC + MC3100_EBC + MobileX_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 54: Real-time capacity (EBC) by system on page 218.

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 64: Software release degradation factors (SWRC) on page 260 lists the software release degradation factors for supported software upgrades.

Table 64: Software release degradation factors (SWRC)

From	Degradation factor (%)
	to Avaya CS 1000 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

From	Degradation factor (%)
	to Avaya CS 1000 Release 7.5
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 7.5	6

Table 65: Ratio of existing processor capacity to new processor capacity (CPTU) on page 261 gives capacity ratio values for supported processor upgrades.

Table 65: Ratio of existing processor capacity to new processor capacity (CPTU)

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

DSP/Media Card calculations

DSP resources are provided by Media Cards. The total DSP/Media Card requirement is the sum of DSP requirements for various functions, which are calculated separately.

- DSP ports for TDS/Conference on page 262
- DSP ports for general traffic on page 263
- DSP ports for major applications on page 264
- Special ACD treatment for nonblocking access to DSP ports on page 265
- Total DSP requirements on page 266
    - General configuration (ACD agent telephones less than 15 percent of total telephones) on page 266
    - Call center application (ACD agent telephones greater than 15 percent of total telephones) on page 266

For reasons explained in the "System capacities on page 193" chapter (see Traffic capacity engineering algorithms on page 216), the Erlang B model is used to calculate DSP port requirements.

For Media Card 32-port cards, the DSP port requirement must be calculated in increments of 32. Table 66: Erlang B and Poisson values, in 32-port increments on page 262 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 267).

Table 66: 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

To obtain the exact number of DSP ports required, use the following formula. Round up to the next integer if the result is a fraction.

Number of DSP ports = (Calculated CCS) ÷ (CCS from Table 66: Erlang B and Poisson values, in 32-port increments on page 262) × (Number of DSP ports for table CCS)

For example, a calculated value of 2430 CCS requires 81 DSP ports to provide a P.01 GoS (2430 ÷ 2891 × 96 = 81). Note that, for Media Card 32-port cards, this implies the use of 3 Media Cards, or 96 ports.

DSP ports for TDS/Conference

In Large Systems, the dual-function TDS/Conference card provides Tone and Digit Switch (TDS) tone services as well as conference circuits.

TDS provides tones to TDM lines and trunks, while IP Phones generate their own tones. Therefore, the demand for TDS services is reduced proportional to any increase in the number of IP Phones in the system. The recommendation of two TDS/Conference cards per network is valid for estimating the service circuit requirement.

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

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, on the assumption that each network group is equipped with two TDS/Conference cards, the default is two conference loops, with a total of 60 channels, per network group. If a particular application requires a different number of conference ports, use the specific number.

The equation to calculate the number of DSP ports the system requires for Conference is:

Equation 1

Number of DSP ports for Conference = (Total number of telephones) $\times P_{IP} \times r_{CON} \times 0.4$

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 ($r_{CON} = 2 \div 28 = 0.07$). The default value of r_{CON} can be changed if circumstances warrant.

Since ports generally have light traffic while channels have heavy traffic, the factor 0.4 is applied in Equation 1 to take account of the high concentration of telephones to channels and adjust for the ratio of ports to channels.

Note that the number of DSP ports for Conference is directly proportional to the system's IP ratio (P_{IP}).

DSP ports for general traffic

There are three steps to calculate the number of DSP ports required for general traffic:

1. Calculate the number of calls that require DSP resources.

   DSP calls (C_{DSP}) = all calls that require a DSP $C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI} + C_{1SIP} + C_{STSD} + C_{TSDS}$ + TDM to TDM using ELC

   For sites where the proportion of ACD agent telephones is less than 15% of the total telephones in the system, C_{DSP} includes all general traffic seeking DSP service.

   Sites where the proportion of ACD agent telephones exceeds 15% of the total telephones in the system are considered to be call centers. For call centers, C_{DSP} is a reduced total that excludes ACD CCS. See Special ACD treatment for nonblocking access to DSP ports on page 265.

2. Convert DSP calls to CCS.

   DSP CCS = $C_{DSP} \times$ WAHT $\div$ 100

3. Using the Erlang B table for P.01 GoS (see Table 66: Erlang B and Poisson values, in 32-port increments on page 262), find the corresponding number of DSP ports required.

Equation 2

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

DSP ports for major applications

For most applications, use the following rules:

- For a pure IP system, provide one DSP port for each application port.
- For a mixed IP and TDM system, calculate the DSP port requirement by multiplying the number of application ports by the fraction of IP calls in the system (the IP ratio, P_{IP}).

Table 67: DSP port requirements for applications on page 264 provides the equations to calculate the number of DSP ports required for each application.

Table 67: DSP port requirements for applications

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

Equation 3

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

Special ACD treatment for nonblocking access to DSP ports

The following section applies for call centers, which are defined as sites where the number of ACD agent telephones exceeds 15% of the total telephones in the system.

Since both Erlang B and Poisson models assume a high ratio of traffic sources to circuits, using the standard estimate of 36 CCS per agent to calculate DSP requirements for a specified GoS tends to result in over-provisioning. For this reason, rather use the fixed rule of one DSP port for each ACD agent telephone requiring a DSP resource, in order to provide nonblocking access between an ACD agent telephone and a DSP. ACD agent telephones require DSP resources only when calls are coming from TDM trunks to IP agent telephones or from Virtual Trunks to TDM agent telephones.

In general, Media Cards are system resources that are available to all traffic sources, including ACD agent telephones and regular phones. Zoning control is the only way to provide nonblocking access to DSP ports for ACD agent telephones only. In a multiple-zone network, each zone is controlled by the Network Routing Service (NRS). When a zone is designated as a private zone for a specific group of ACD agent telephones, service requests from outside the protected zone to a designated group of DSP resources are denied.

Assuming that zoning control has been established and that a group of Media Cards can be reserved for the exclusive use of ACD agents, recalculate the number of DSP ports required for general traffic excluding ACD agent CCS, and then add in DSP ports required for the ACD agent telephones. The steps are as follows:

1. Calculate system CCS excluding ACD agents. Since system CCS is two-way traffic, the traffic associated with both incoming and outgoing trunks terminating on ACD agents must be removed:

   Reduced system CCS = Total system CCS (T_{CCS}) – [2 × (Number of ACD agent telephones) × CCS/agent]

2. Recalculate the intraoffice ratio (R_I), IP ratio (P), Virtual Trunk ratio (V), and other ratios to reflect the new distribution of call types without ACD traffic. (See Table 60: Resource calculation parameters on page 248 for the equations to calculate the ratios.

3. Use the reduced system CCS and new ratios to calculate calls requiring DSP and Virtual Trunk resources. (See Traffic equations and penetration factors on page 252 for the detailed calculations for the different call types.)

4. Convert DSP calls to CCS.

   DSP CCS = C_{DSP} × WAHT ÷ 100

5. Using the Erlang B table for P.01 GoS (see Table 66: Erlang B and Poisson values, in 32-port increments on page 262), find the corresponding number of DSP ports required (for general traffic without ACD agents).

   Equation 2a

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

6. Calculate the DSP requirement for ACD agent telephones. A DSP port is needed only when calls are coming from TDM trunks (ratio 1 – V) to IP agent telephones or from Virtual Trunks (ratio V) to TDM agent telephones.

   Equation 4

   Number of DSP ports = (Number of IP ACD agent telephones) × (1 – V) + (Number of TDM ACD agent telephones) × V

Total DSP requirements

General configuration (ACD agent telephones less than 15 percent of total telephones)

Total number of DSP ports = DSP ports for TDS/Conference on page 262 + DSP ports for general traffic on page 263 + DSP ports for major applications on page 264

Call center application (ACD agent telephones greater than 15 percent of total telephones)

Total number of DSP ports = DSP ports for TDS/Conference on page 262 + Special ACD treatment for nonblocking access to DSP ports on page 265 + DSP ports for major applications on page 264

Virtual Trunk calculations

For reasons explained in the "System capacities on page 193" chapter (see Traffic capacity engineering algorithms on page 216), the Poisson model is used to calculate trunk requirements.

Table 66: Erlang B and Poisson values, in 32-port increments on page 262 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 66: Erlang B and Poisson values, in 32-port increments on page 262) × (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 ACD treatment for nonblocking access to DSP ports on page 265.

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}) = $VT_{CCS} \times v_S$ H.323 Virtual Trunk CCS (HVT_{CCS}) = $VT_{CCS} \times v_H$

5. Using the Poisson table for P.01 GoS (see Table 66: Erlang B and Poisson values, in 32-port increments on page 262 or Trunk traffic Erlang B with P.01 Grade-of-Service on page 409), 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 may 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 may 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 292.

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 will render 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 which 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 = $VT_{CCS} \div 36$

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 the bandwidth table and for more information about 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 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 the

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 68: Signaling Server algorithm constants on page 270 defines the constants you use in the Signaling Server algorithm.

Table 68: 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.
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 Common Server = 300 000	Software limit.
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	Hardware limit.

Algorithm Constant	Description	Limit	Notes
		COTS2 = 80 000 Common Server = 80 000	
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.
$SIPL_{SL}$[type_index]	SIP Phone limit	CP PM = 1800 CP DC = 3700 COTS1 = 1800 COTS2 = 3700 Common Server = 3700	Software limit, includes SIP DECT

Algorithm Constant	Description	Limit	Notes
$SIPL\_Vtrk_{SL}$[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.
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.

Algorithm Constant	Description	Limit	Notes
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
$MSCS_{SL}$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	Hardware limit, calls per hour.

Algorithm Constant	Description	Limit	Notes
		Common Server = 120 000	
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 IBM x3350 = 800 DELL R300 = 700 HP DL360 G7 = 800	Software limit CP PM and COTS1 not supported
MASC$_{SL}$[type_index]	MAS session call rate limit	CP DC = 2000 COTS2 = 6000	Hardware limit for cph

Algorithm Constant	Description	Limit	Notes
		Common Server = 6000	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 69: Signaling Server algorithm user inputs on page 275 describes the user inputs you use in the Signaling Server algorithm.

Table 69: 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 provisioned in this node)
SSDB[type_index]	PD/RL/CL required	enter / derived	= a if not required = b if shared = c if standalone

Algorithm user input	Description	Value	Notes
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
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

Algorithm user input	Description	Value	Notes
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 70: Signaling Server algorithm variables on page 277 describes the variables you use in the Signaling Server algorithm.

Table 70: 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)
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).

Algorithm variable	Description	Value	Notes
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.
CallRate	Total call rate calculated for Co-resident Signaling Server	calc	Initialize = 0
NumOfCo-res	Number of Co-resident Signaling Server applications	calc	Initialize = 0
NRS_Co-res	NRS Co-resident	calc	True or False
TPS_Co-res	TPS Co-resident	calc	True or False
PD_Co-res	PD Co-resident	calc	True or False

Algorithm variable	Description	Value	Notes
H323_Co-res	H.323 Co-resident	calc	True or False
SIP_Co-res	SIP Co-resident	calc	True or False
TR87_Co-res	TR87 Co-resident	calc	True or False
SLG_Co-res	SLG Co-resident	calc	True or False
MSC_Co-res	Media Session Controller Co-resident	calc	True or False
MSC_sessions	Number of MSC sessions	calc	
MSCC	MSC call rate	calc	Calculated MSC call rate for the number of required sessions
ELC_{VT}	Number of ELC SIP trunks	calc	Set to ELC ISM value
STBC[type_index]	SIP Trunk Bridge calls per hour	calc	Calculated STB calls per hour
SSSTBR[type_index]	STB Signaling Server calculation	calc	Calculated number of real Signaling Servers for SIP Trunk Bridge
SSSTBW[type_index]	STB Signaling Server requirements	calc	Calculated whole number of Signaling Servers for SIP Trunk Bridge, including alternate
SSMSCR[type_index]	MSC Signaling Server calculation	calc	
SSSTBW[type_index]	MSC Signaling Server requirements	calc	
SSMASW[type_index]	MAS Signaling Server requirements	calc	
SSMASR[type_index]	MAS Signaling Server calculation	calc	

Algorithm variable	Description	Value	Notes
SSMASRed[type_index]	Number of MAS redundant SS required	calc	
HS_SS[type_index]	HS unique Signaling Server count	calc	
SST[type_index]	Total count of required Signaling Servers	calc	

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.

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

1. Signaling Server for Personal Directory/Callers List/Redial List database (SSDB) on page 280
2. Network Routing Service calculation (SSNR) on page 281
3. Terminal Proxy Server calculation (SSTR) on page 283
4. H.323 Gateway calculation (SSHR) on page 284
5. SIP Gateway calculation (SSSR) on page 285
6. SIP CTI/TR87 Calculation on page 285
7. Signaling Server Co-resident calculations on page 286
8. SIP Line Gateway calculations on page 289
9. SIP DECT Server calculations on page 290
10. Total Signaling Servers (SST) on page 291

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).

1. 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

2. 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) or 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 71: SIP mode factors on page 282.

Table 71: SIP mode factors

type_index	SIP_Redirect_Factor	SIP_Proxy_Factor
CP PM	2	4
CP DC	1.5	3
HP (HP DL320-G4)	1.5	3
IBM1 (IBM x306m)	1.5	3
IBM2 (IBM x3350)	1.67	2.5
DELL2 (Dell R300)	1.67	2.5
HP (HP DL360 G7)	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_l[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 = $VT_{323} \times CCS \times 100 \div WAHT$ SIP calls = $VT_{SIP} \times CCS \times 100 \div WAHT$

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) $NRC_{NET} = VT_{NET} \times$ 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.

3. Terminal Proxy Server calculation (SSTR)

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

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

}

}

4. H.323 Gateway calculation (SSHR)

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

SSHR[type_index] = larger of:

{

a	$HVT[type\_index] \div HVT_{SL}[type\_index]$	number of trunks (software limit)
b	$HC_{VT}[type\_index] \div HVTC_{HL}[type\_index]$	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

}

}

5. SIP Gateway calculation (SSSR)

SC_{VT} = (SVT_{CCS} × 100) ÷ WAHT

SSSR[type_index] = larger of:

{

a	SVT[type_index] ÷ SVT_{SL}[type_index]	number of trunks (software limit)
b	SC_{VT}[type_index] ÷ $SVTC_{HL}$[type_index]	calls per hour (hardware limit)
c	SVT[type_index] > 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

}

}

6. 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

}

}

7. 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)

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;

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 > 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};

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

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)

Then

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

}

Else { Call rate exceeds the Co-res limit

TNRC = NRC;

$TC_{UIP} = C_{UIP}$;

$THC_{VT} = HC_{VT}$;

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

$TC_{SIP} = C_{SIP}$;

DO

{

large = largest of (TNRC, TC_{UIP}, THC_{VT}, TSC_{VT}, T_{SIP}); 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;
         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 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)
Then     SST[type_index] = SST[type_index] + 1;
         }
         }
}
}
```

8. SIP Line Gateway calculations

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	$SIPL \div SIPL_{SL}$[type_index]	number of phones (software limit)
b	$C_{SIP} \div SIPLC_{HL}$[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;	
	}	

9. SIP DECT Server calculations

The SIP Line Gateway (SLG) calculation includes SIP DECT. No independent SIP DECT servers required.

SS_DECT = 0

10. 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]:

See Signaling Server calculation on page 303 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 72: RSF algorithm constant and variable definitions on page 291 defines the terms used in the calculations.

Table 72: 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.

Algorithm term	Description	Value	Notes
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 may be a high intraoffice ratio in a call center, or few trunks but a high interoffice ratio. In these cases, the resulting calculations in the EC tool will 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 may occur

- Virtual Trunks on page 292
- Line and trunk traffic on page 293

Virtual Trunks

When the VT number input by the user differs significantly from the calculated VT number (more than 20% difference), the EC tool will use 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 will repeat the calculation cycle up to six times. This recalculation looping is built into the EC and occurs automatically, with no user action required. At the end of the recalculation cycle, the user has the choice of using the original input VT number or the final calculated VT number in the configuration.

The user inputs about the number of H.323 Virtual Trunks and SIP Virtual Trunks are a function of other parameters in the system. For example, the number of Virtual Trunks required will be 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 preengineering has not provided consistent ratios and CCS, the VT input numbers will 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 HVT_{CCS}, and between SVT and SVT_{CCS}, is the deviation between input data and calculated value.

After the automatic EC recalculations, if the discrepancy between the input VT number and the final calculated number is still significant (more than 20%), follow the recommendations for reducing line and trunk traffic imbalance (see Line and trunk traffic on page 293). Adjusting the number of Virtual Trunks and trunk CCS alone, without changing the intraoffice ratio or its derivatives, may never get to a balanced configuration.

Line and trunk traffic

At the end of the algorithm calculation, if the line and trunk CCS are significantly imbalanced (more than 20% difference), the EC tool will generate a warning message. The user can choose whether to change input data and rerun the calculation to have a better balanced system. The recalculation loop starts from the point of entering configuration inputs at the GUI.

Use the following formula to obtain the calculated line CCS to compare against user input, in order to determine the size of the deviation.

Calculated line CCS (LC_{CCS}) = $(C_{SS} + C_{ST} + C_{TS}) \times WAHT \div 100$

The difference between L_{CCS} and LC_{CCS} is the imbalanced line CCS.

Similarly, use the following formula to obtain the calculated trunk CCS to compare against user input, in order to determine the size of the deviation.

Calculated total trunk CCS (TC_{CCS}) = $(C_{TT} + C_{ST} + C_{TS}) \times WAHT \div 100$

The difference between T_{TCCS} and TC_{CCS} is the imbalanced trunk CCS.

Because the calculated CCS factor in traffic ratios, line and trunk CCS can be significantly imbalanced if these ratios are inconsistent. For example, if the intraoffice, incoming, and outgoing ratios are based on contradictory assumptions, the calculated line CCS may be much higher than the number of trunks can absorb.

Table 73: Tips to reduce traffic imbalances on page 294 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 73: 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 will have 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 which 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 1000M Large 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.)
- Avaya 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 62: Real-time factors on page 256.

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 263. 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 = 0.567
- 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} × R_I = 19578 × 0.25 = 4895 calls

        i. Intraoffice UNIStim IP to UNIStim IP calls (C_{2IP}) = C_{SS} × P_U × 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} × 2 × P_U × (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}) = $2 \times C_{TT} \times V \times (1 - V)$ = 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}) = $C_{TT} \times (1 - V) \times (1 - V)$ = 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}) = $C_{TT} \times V^2 \times V_H \times V_S \times 2 \times 2$ = 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}) = $T_{CALL} \times O$ = 19578 × 0.12 = 2349 calls

i. UNIStim IP to VT calls (C_{STIV}) = $C_{ST} \times P_U \times V$ = 2349 × 0.346 × 0.549 = 446 (require VT) P_UIPtoVT = 446 ÷ 19578 = 0.02

ii. UNIStim IP to TDM trunk calls (C_{STID}) = $C_{ST} \times P_U \times (1 - V)$ = 2349 × 0.346 × (1 – 0.549) = 367 (require DSP) P_UIPtoTr = 367 ÷ 19578 = 0.02

iii. TDM telephone to VT calls (C_{STDV}) = $C_{ST} \times (1 - P_{IP}) \times (V)$ = 2349 × (1 – 0.567) × 0.549 = 558 (require DSP, VT) P_LtoVT = 558 ÷ 19578 = 0.03

iv. TDM to TDM calls (C_{STDD}) = $C_{ST} \times (1 - P_{IP}) \times (1 - V)$ = 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}) = $C_{ST} \times P_S) \times V$ = 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}) = $C_{ST} \times P_S) \times (1 - V)$ = 2349 × 0.221 × (1 - 0.549) = 234 (require DSP, no VT) P_SIPtoTr = 234 ÷ 19578 = 0.01

d. Terminating/incoming calls (C_{TS}) = $T_{CALL} \times I$ = 19578 × 0.60 = 11747 calls

i. VT to TDM telephone calls (C_{TSVD}) = $C_{TS} \times V \times (1 - P_{IP})$ = 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}) = (0.3 × 0.13) + (0.07 × 1.08) + (0.05 × 0.03) + (0.015 × 1.83) + (0.006 × 2.07) + (0.008 × 1.60) + (0.02 × 1.33) + (0.02 × 2.00) + (0.03 × 1.75) + (0.01 × 2.30) + (0.14 × 1.84) + (0.11 × 1.87) + (0.09 × 2.13) + (0.12 × 1.40) + (0.01 × 2.90) + (0.04 × 1.52) + (0.05 × 1.48) + (0.01 × 3.12) + (0.01 × 3.58) + (0.07 × 2.30) + (0.06 × 2.27) = 1.705

- Calculate the System EBC SEBC = (T_{CALL} × (1 + PF + Error_term)) = 19578 × (1 + 1.705 + 0.25) = 57857

Real-time calculation with major applications

- ACD agent calls without Symposium C_{ACD} = [(L_{ACD} + L_{IPACD}) × 100 ÷ AHT_{AGT}] = (5610 × 1650) × 100 ÷ 180 = 4033
- Calculate the impact of other major features and applications.

  You can use Table 74: Feature and application EBC calculation on page 299 to calculate the EBC for your features and applications.

Table 74: Feature and application EBC calculation

Feature or application	EBC calculation forumula	EBC
ACD	ACDEBC = C_{ACD} × (1 - %Symposium) × f_{ACD}	= 4033 × 0.1 × 0.13 = 52

Feature or application	EBC calculation forumula	EBC
Symposium	SymposiumEBC = %Symposium × C_{ACD} × f_{SYM}	= 4033 × 0.9 × 5.74 = 20836
CallPilot	CallPilotEBC = (CP1 + CP2) × 100 ÷ AHT_{CP} × f_{CP}	= (936 + 624) × 100 ÷ 40 × 1.66 = 6474
Incoming CDR	IncomingCDR_EBC = C_{TS} × f_{CCDR}	= 11747 × 0.32 = 3759
Outgoing CDR	OutgoingCDR_EBC = C_{ST} × f_{OCDR}	= 2349 × 0.32 = 752
Tandem CDR	TandemCDR_EBC = C_{TT} × f_{TAN}	= 587 × 0.44 = 258
Integrated Conference Bridge	MICB_EBC = number of Integrated Conference Bridge ports × $Appl_{CCS}$ × 100 ÷ AHT_{MICB} × f_{MICB}	= 16 × 26 × 100 ÷ 1800 × 1.56 = 37
Integrated Recorded Announcer	MIRAN_EBC = number of Integrated Recorded Announcer ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIRAN} × f_{MIRAN}	= 12 × 26 × 100 ÷ 90 × 0.63 = 218
Integrated Call Director	MIPCD_EBC = number of Integrated Call Director ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIPCD} × f_{MIPCD}	= 12 × 26 × 100 ÷ 60 × 0.63 = 328
Integrated Call Announcer	MIPA_EBC = number of Integrated Call Announcer ports × $Appl_{CCS}$ × 100 ÷ AHT_{MICA} × f_{MICA}	= 8 × 26 × 100 ÷ 180 × 0.57 = 66
Hospitality Integrated Voice Service	MIVS_EBC = number of Hospitality Integrated Voice Service ports × $Appl_{CCS}$ × 100 ÷ AHT_{MIVS} × f_{MIVS}	= 8 × 26 × 100 ÷ 90 × 0.57 = 132
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 +	= 41332

Feature or application	EBC calculation forumula	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	

New system real-time usage

Compare the total system EBC with the CPU rated capacity to determine the processor utilization.

RTU = (SEBC + FAEBC) ÷ BHCC × 100 = (57857 + 41332) ÷ 840 000 × 100 = 11.8 %

In this example, CPU utilization, including application and feature impact, is 11.8 %. 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 64: Software release degradation factors (SWRC) on page 260 and Table 65: Ratio of existing processor capacity to new processor capacity (CPTU) on page 261:

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 (29.8 + 11.8) = 41.6% .

DSP calculation for general traffic (no applications)

DSP calls (C_{DSP}) = $C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI} + C_{1SIP} + C_{STSD} + C_{TSDS}$= 9650

DSP CCS = C_{DSP} × WAHT ÷ 100 = 9650 × 128 ÷ 100 = 12352 CCS

Refer to an Erlang B table (with P.01 GoS) to find the corresponding number of ports, or use the following formula:

Number of DSP ports = DSP CCS ÷ 6192 × 192 = 384

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:

DSP channels for conference ports = Total_sets × P_{IP} × r_{CON} × 0.4 = 4208 × 0.567 × 0.07 × 0.4 = 67

DSP calculation for Features and applications

Feature or application	Number of ports	DSP
Integrated Conference Bridge	16	9.07
MIRAN	12	6.80
Integrated Call Director	12	6.80
Integrated Call Assistant	8	4.54
Integrated Voice Services	8	4.54
BRI	8	4.54
Agent greeting	4	2.27
CallPilot	60	34.02
Total	73	

Feature or application	Number of ports	DSP
The general formula for calculating DSP for features and applications is: DSP = number of ports × P_{IP}		

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.

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}) = $C_{VT} \times V_H$ = 8184 × 0.67 = 5484

SIP VT calls (SC_{VT}) = $C_{VT} \times V_S$ = 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 = CPPM]

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 for Personal Directory/Callers List/Redial List

   SSDB = b

   The database share limit for type_index = CPPM is 2000 IP Phones.

2. Network Routing Service calculations

   Dedicated NRS not required, nrs_type_index = type_index = CPPM

   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[CPPM] = 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[CPPM] = 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

3. Terminal Proxy Server calculations

Calculate TPS call rate:

$C_{UIP} = C2_{UIP} \times 2 + C1_{UIP} + C2_{SIPUIP} + C_{STIV} + C_{STID} + C_{STVI} + C_{STDI}$

$C_{UIP} = 8267$

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[CPPM] = 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

}

}

4. H.323 Gateway calculations

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

$= (8400 \times 100) \div 128$

$= 6562$

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[CPPM] = 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

}

}

5. SIP Gateway calculations

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

$= (4200 \times 100) \div 128$

$= 3500$

SSSR[type_index] = larger of:

{

a SVT[type_index] ÷ SVT_{SL}[type_index] = 150 ÷ 1800 = 0.08

b SC_{VT}[type_index] ÷ $SVTC_{HL}$[type_index] = 3500 ÷ 27 000 = 0.13

c SVT > $VRTK_{SL}$[type_index] then 1 else 0 150 > 800 evaluates false = 0

}

SSSR[CPPM] = 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

}

}

6. SIP CTI/TR87 calculations

No SIP CTI/TR87 specified.

7. Signaling Server Co-resident calculations

Determine if any Signaling Server applications can coreside 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[CPPM] = 1 + 1 = 2;

Since one of the following evaluates true; SST[CPPM] = 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;

}

}

}

8. 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[CPPM] = 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[CPPM] = 2 × 2 = 4

SST[type_index] = SST[type_index] + SSSLGW; SST[CPPM] = 3 + 4 = 7

9. SIP DECT calculations

No SIP DECT Phones required.

10. 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 [CPPM] = type_index [CPPM] 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 = $[(240 + 120) \times 28] \div 36 = 280$ erlangs

Chapter 15: Application engineering

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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, see feature documentation in http://www.avaya.com/support.

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 75: CP PIV packet throughput drop at 10 percent CPU utilization on page 312 provides an example of the CP PIV performance drop with increasing rule depth.

Table 75: 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 telephone configured to access Multimedia Communication Server (MCS) 5100 multimedia applications through a Session Initiation

Protocol (SIP) Virtual Trunk. An Avaya Communication Server 1000M (Avaya CS 1000M) system equipped with a CP PIV processor has sufficient real-time capacity to support the Converged Desktop application on all telephones in the system.

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 will 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.

SIP Access Port and PCA Licenses are included with the purchase of MCS 5100 Converged Desktop telephones. Customers do not usually need to purchase incremental software licenses.

Calculating SIP access port and PCA requirements

Table 76: SIP port and PCA requirements for Converged Desktop (with P.01 GoS) on page 314 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 will 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 76: SIP port and PCA requirements for Converged Desktop (with P.01 GoS) on page 314 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, extrapolate from the closest number.

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 76: 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

# CD Users		% voice traffic carried by SIP trunk											
		0	**5**	**10**	**15**	**20**	**25**	**30**	**35**	**40**	**45**	**50**	**100**
	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
	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

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
250	SIP CCS	125.0	181.2	237.5	293.8	350.0	406.2	462.5	518.8	575.0	631.2	687.5	1250.0
	SIP port	9	12	14	16	19	21	23	25	27	29	31	50
	PCA	9	9	9	9	9	9	9	9	9	9	9	9
300	SIP CCS	150.0	217.5	285.0	352.5	420.0	487.5	555.0	622.5	690.0	757.5	825.0	1500.0
	SIP port	10	13	16	19	21	24	26	28	31	33	36	58
	PCA	10	10	10	10	10	10	10	10	10	10	10	10
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
	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

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
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
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

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
	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
	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

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
	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
11 000	SIP CCS	5500	7975	10 450	12 925	15 400	17 875	20 350	22 825	25 300	27 775	30 250	55 000
	SIP port	183	257	331	404	477	549	621	693	769	842	916	1651
	PCA	183	183	183	183	183	183	183	183	183	183	183	183
12 000	SIP CCS	6000	8700	11 400	14 100	16 800	19 500	22 200	24 900	27 600	30 300	33 000	60 000
	SIP port	198	279	359	439	518	597	675	754	837	917	997	1799
	PCA	198	198	198	198	198	198	198	198	198	198	198	198
13 000	SIP CCS	6500	9425	12 350	15 275	18 200	21 125	24 050	26 975	29 900	32 825	35 750	65 000
	SIP port	213	301	387	473	559	644	729	819	905	992	1079	1948
	PCA	213	213	213	213	213	213	213	213	213	213	213	213
14 000	SIP CCS	7000	10 150	13 300	16 450	19 600	22 750	25 900	29 050	32 200	35 350	38 500	70 000
	SIP port	228	322	415	508	600	691	787	880	974	1067	1161	2096
	PCA	228	228	228	228	228	228	228	228	228	228	228	228
15 000	SIP CCS	7500	10 875	14 250	17 625	21 000	24 375	27 750	31 125	34 500	37 875	41 250	75 000
	SIP port	243	344	443	542	640	738	842	942	1042	1142	1243	2245

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
	PCA	24 3	243	243	243	243	243	243	243	243	243	243	243
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 Converged Office feature combines the business-grade telephony of the Avaya Communication Server 1000 (Avaya CS 1000) with the real-time multimedia communication and the remote call control provided by Microsoft® Office Communications Server (OCS) 2007. 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 77: 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 78: Traffic figures on page 322 shows traffic in CCS and number of ports calculated based on Poisson formula at P.01 Grade of Service.

Table 78: 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

Traffic (CCS)	Traffic (Erlang)	#Ports
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

Traffic (CCS)	Traffic (Erlang)	#Ports
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 services requirements

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 61: Capacity example on page 325).

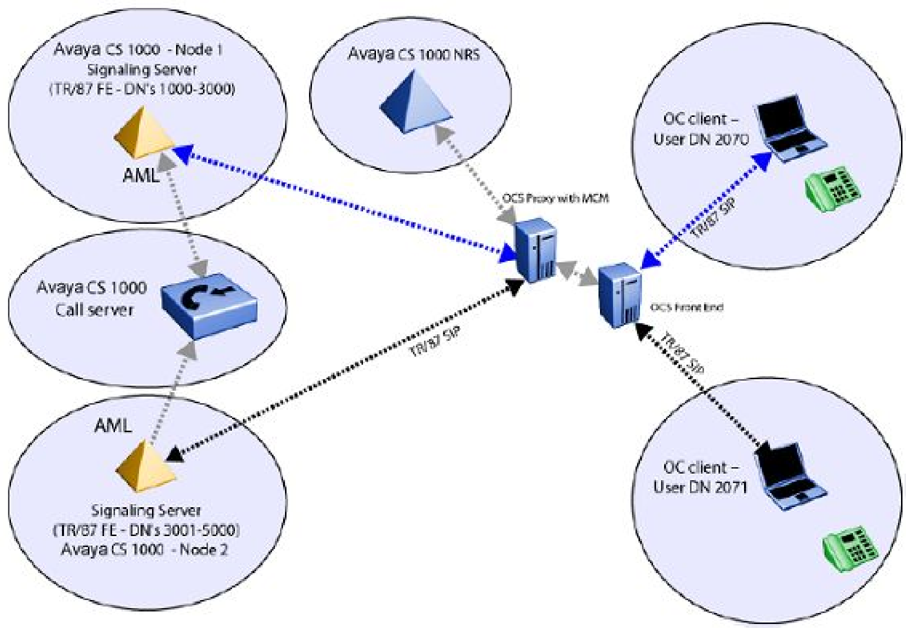

Figure 61: Capacity example

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 Call Server supports 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 12 000 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. DSPs are system resources on a Communication Server 1000M system.

MOBX requires PRI trunks for DTMF usage on a Communication Server 1000M system. You must provide one DSP for each trunk in the same physical group as the PRI trunks. MOBX can only use DSP ports in the same group as the MOBX PRI trunks. Avaya recommends you to configure the all the DSP for MOBX calls in private bandwidth zone 0. No other entities should be in bandwidth zone 0. Do not configure Virtual Trunks, IP Phones, or other DSP resources in bandwidth zone 0. In this configuration the MOBX PRI trunks can always access the DSP from bandwidth zone 0 for DTMF monitoring.

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 × T) / CHT

Where NC = number of calls

T = traffic in CCS

CHT = 150 (average call hold time in seconds)

DTR CCS = (HT × NC) / 100

Where HT = 6 (DTR average hold time in seconds)

Use your DTR CCS calculated call value and see Table 126: Digitone receiver load capacity 6 to 15 second holding time on page 417 to find the number of DTRs required under the 6 second column.

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 120: Trunk traffic Poisson 1 percent blocking on page 411).

MOBX PRI engineering example

1000 MOBX users

5 CCS busy hour call attempts (BHCA)

Therefore, 1000 × 5 = 5000 CCS

The table indicates 5000 CCS requires 168 trunks and 168 DSPs, a one to one relationship between DSP and DTR trunks.

Multi-purpose Serial Data Link

Prior to the introduction of the Multi-purpose Serial Data Link (MSDL) card, a system could support a total of 16 I/O ports. Now, a system can support up to 16 MSDL cards, each of which can be flexibly configured to support combinations of D-channel (DCH), Application Module Link (AML), Command Status Link (CSL), and Serial Data Interface (SDI) on 4 ports, for a total of 64 I/O ports.

This section provides guidelines to help the user engineer the MSDL. This section contains information about the following topics:

- MSDL engineering considerations on page 328
- MSDL architecture on page 329
- D-channel on page 330

- Application Module Link (AML) on page 339
- Serial Data Interface (SDI) on page 342
- MSDL engineering procedure on page 344
- Examples on page 349

MSDL engineering considerations

These 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 MSDL. At this point the recovery mechanism becomes essential. The mechanism should be graceful, not requiring manual intervention, and should provide as much diagnostic information as possible, to help isolate the root cause of the problem. Refinements and improvements to the recovery mechanisms have been introduced over various software releases.

The D-channel expansion feature increases the number of I/O addresses allowable for D-channel application to 16 per network group and 256 per system. The number of nonD-channel applications is 16 per system (or 64 if all MSDLs are used).

The limit of 256 D-channels is a theoretical limit. Avaya recommends the following limits in practice:

- For office/commercial applications: In a fully equipped 8-group system, the optimal configuration in terms of port capacity and trunking percentage (15%) requires about 112 D-channels, assuming one D-channel per T1. The same optimal configuration can be reached with fewer D-channels in E1 applications.
- For Call Center applications: To achieve a trunk to agent ratio of 1.5:2400, deploy about 144 D-channels in T1 applications. The optimal configuration can be reached at 136 D-channels in E1 applications.

The current MSDL card is used for D-channel expansion. When configuring multiple D-channels on a card, strictly follow the MSDL engineering guidelines. As long as feature penetration is accounted for in the system real time engineering model, D-channel expansion has no direct impact on Core Processor (CP) capacity.

Engineering the MSDL requires an understanding of the end-to-end performance characteristics of the system. Outgoing messages originate from the system CP, are passed to the MSDL, and travel across the appropriate link to the destination. In equilibrium, or over a relatively long period of time (i.e. on the order of several minutes), the system cannot generate messages faster than the MSDL processor can process them, than the link can transmit them, or than the destination can process them. Otherwise, messages will build up at the bottleneck and will eventually be lost. The entity with the lowest capacity will be the system bottleneck. For very short periods of time, however, one or more entities may 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.

MSDL architecture

The MSDL processor is a 68020 processor. The MSDL 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 MSDL/MISP 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 MSDL 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 CP from excess messages. To account for the new, faster processors, the thresholds have been changed so that MSDL304 is printed if 100 messages in 2 seconds is exceeded, MSDL305 is printed if 200 messages in 2 seconds is exceeded, and MSDL306 is printed and the card is disabled if 300 messages in 2 seconds is exceeded. In both cases Background Audit will bring the MSDL back up if no problems are found. The Port Overload Counter is introduced. If the incoming messages on a single port exceed 200 messages in 2 seconds, the port will be locked out, and an MSDL_port_overload message will be printed. Manual intervention is required to clear the overloaded port. This feature prevents a single port from locking up the whole MSDL card.

Several software tasks exist on the MSDL. Layer 1 message processing operates at the highest priority. If the link is noisy, Layer 1 processing may starve the Layer 2 and Layer 3 processing tasks, resulting in buffer overflows. If such a problem is suspected, the Protocol Log (PLOG) should be examined. PLOG reporting is requested in LD 96, as described in the *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 MSDL, so the MSDL performs some functions previously performed by the system core processor, thus reducing the capacity on the MSDL. These interfaces will be 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 which is shared by all D-channels.

It is possible to define overload thresholds for R20+ interfaces on a per-D-channel basis. 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 on a per D-channel basis, 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 will resume accepting new calls at the end of the third time interval. This flexibility allows the user to regulate the MSDL processing required by a specific R20+ DCH port. Note that 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 centi-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 MSDL is able to support basic call processing messages for 4 D-channels under normal operation (see Table 79: Steady-state requirements and capacities per D-channel (outgoing and incoming) on page 331).

Table 79: Steady-state requirements and capacities per D-channel (outgoing and incoming)

System	Requirement msg/sec	System CP capacity msg/sec	MSDL capacitymsg /sec	Link capacity msg/sec	Comment
68060 CP	13(T1)/16(E1)	161	87	212 input 212 output	Limited by traffic requirements
68060E CP	13(T1)/16(E1)	242	87	212 input 212 output	Limited by traffic requirements

Peak analysis

When there is a link re-start, 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 MSDL, exceeding MSDL 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 previous one because it is sending out D-channel messages which do not involve call processing. MSDL and Link capacities are also higher because, for equilibrium analysis, some capacity is reserved for peaking.

Table 80: Peak requirements per D-channel (outgoing) on page 331 illustrates 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 MSDL.

Table 80: Peak requirements per D-channel (outgoing)

System	Burst Size	System capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
68060 CP	382(T1)/ 480(E1)	410	113	276 output	MSDL is bottleneck
68060E CP	382(T1)/ 480(E1)	615	113	276 output	MSDL is bottleneck

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 MSDL cards so that message bursts are not being sent to four ports on the same MSDL 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 and allows the user to configure the number of Status Enquiry messages sent within 128 msec on a per D-channel basis. The parameter, SEMT, is configured in LD 17, and can 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 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 may impact the performance of the MSDL 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.

The B-channel Overload Control feature is introduced to address 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 configured in LD 16, and falls in the range 0 to 4000 msec.

ISDN Signaling Link (ISL) network

In an 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 81: ISL link capacities on page 332). 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 330.

Table 81: 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)

Modem baud rate	Link capacity (msgs/ sec)	B-channels that can be supported
19 200	58 input 58 output	382(T1)/480(E1)

For the baud rates listed in Table 81: ISL link capacities on page 332, the link will be the limiting constraint. The potential peak traffic problems described in Peak analysis on page 331 apply here as well, to an even greater extent since the rate mismatch between the system and the system bottleneck, now the link instead of the MSDL, is greater. To minimize the risk, set the baud rate as high as possible.

Virtual Network Services (VNS) network

The concepts mentioned in ISL networks also applies to VNS networks. Up to 4000 VNS DNs (VDN) are supported.

D-channel bit rate

These guidelines provide the basis for engineering the 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 will backlog.

If the customer is having problems networking calls during high traffic then the D-channel may 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 will fail to network and many PRI/VNS/DCH messages will be 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.

It is recommended 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 will 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 in which each agent node is connected to a tandem node (see Table 82: Steady-state requirements and capacities per D-channel with staggered timers (outgoing and incoming) on page 335). 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.

Table 82: Steady-state requirements and capacities per D-channel with staggered timers (outgoing and incoming)

System	Requirement (msg/sec)	Meridian 1 CP capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
68060 CP	5	161	87	212 input 212 output	Limited by traffic requirement s
68060E CP	5	242	87	212 input 212 output	Limited by traffic requirement s

For this same network, assume now that the timers in the Routing Table are not staggered; instead, Logical Call Requests are broadcast to the four 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 will increase significantly, to the values provided in Table 83: Steady-state requirements and capacities per D-channel with immediate broadcast of Logical Call Requests (outgoing and incoming) on page 335, using the following calculations:

- 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

Table 83: Steady-state requirements and capacities per D-channel with immediate broadcast of Logical Call Requests (outgoing and incoming)

System	Requirement (msg/sec)	Meridian 1 CP capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
68060 CP	14.5	161	87	212 input 212 output	Limited by traffic requirements
68060E CP	14.5	242	87	212 input 212 output	Limited by traffic requirements

Peak analysis

When the CRQS threshold is reached, the target queue will broadcast messages to the source ACD DNs informing them that it will 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 may have queued at source ACD DNs, resulting in a burst of Logical Call Request messages once the DN becomes available.

Unlike the PRI network case, there is no specific worst case scenario for peakedness. The examples in Table 84: Peak requirements for NACD messages (outgoing and incoming) on page 336 are based on a five-node network, where each node has three source ACD DNs.

Table 84: Peak requirements for NACD messages (outgoing and incoming)

System	Burst size		Meridian 1 capacity(msg/sec)	MSDL capacity (msg/ sec)	Link capacity (msg/sec)		Comment
	Outgoing	Incoming			Output	Input	
68060 CP	12	40	410	113	258	258	MSDL is bottleneck
68060E CP	12	40	615	113	258	258	MSDL is bottleneck

If CRQS values are set high, many messages will be exchanged, with the network emulating a single virtual queue. If the CRQS values are lowered, fewer Call Requests will be sent across the network, however, average source delays may be increased. If FCTH levels are set too low, target nodes can ping pong 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 may be inundated with Logical Call Request messages once it becomes available. CRQS is configurable for the range [0, 255] while FCTH is configurable for the range [10, 100]. Since the impact of these parameters is so configuration dependent, it is beyond the scope of this document to make recommendations on how to configure them. They should 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, and 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, 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 will be 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: $p_i = n_i/N$
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: $C_{ij} = A_i \times p_j$
4. Calculate the total calls routed (T, expressed in CPH) between each pair of sites: $T_{ij} = T_{ji} = C_{ij} + C_{ji}$
5. Apply Erlang B to each T_{ij}, i < j, to get the number of required trunks between sites i and j (L_{ij}).

   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)

   See Trunk traffic Erlang B with P.01 Grade-of-Service on page 409 for values for Erlang B.

Parameter settings

The following are parameters that can be configured in LD 17 for Meridian 1 D-channels. They are listed with their input range and default value 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 Meridian 1 CP to the MSDL card. The more that are created, the deeper the buffering. Normally a message created in a buffer is sent to the MMIH (Meridian MSDL Interface Handler) and copied into the ring. If the ring is flow controlled, the message occupies a buffer until it can be sent. For systems with extensive D-channel messaging, such as call centers using NACD, the parameter should be configured at 127. For other systems with moderate levels of D-channel messaging, OTBF should be configured 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 which are located on two different MSDLs, and 2 back-up D-channels, the total number of B-channels is (7 × 24) - 4 = 164. OTBF should 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 MSDL's Layer 2 LAPD will wait before it retransmits a frame. It if doesn't receive an acknowledgment from the far end for a given frame before this timer expires, it will retransmit 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, may require higher values.
3. T203 2 - (10) - 40: Link Idle Timer (units of seconds)

   This timer defines how long the Layer 2 LAPD will wait without receiving any frames from the far end. If no frames are received for a period of T203 seconds, the Layer 2 will send 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 will delay the system from detecting that the far end has dropped the link and initiating the recovery process. The value should be higher than T200. It should 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 will resend a frame if it doesn't 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 will retransmit 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 may 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 Meridian 1 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 will send 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 will receive 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 allows the user to regulate the MSDL 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 an MSDL. To limit the real time utilization of a single R20+ DCH port to (1 ÷ n) of the real time capacity of the MSDL, 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 1/3 of the processing capacity of the MSDL, 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 MSDL Layer 3 software. At the end of the third 5 second interval, Layer 3 will resume accepting incoming call requests.

Application Module Link (AML)

The Application Module Link (AML) provides the connection between the system and the CCR, Meridian Link, or Meridian 911 module. The current maximum speed for the link is 19200 baud. CCR is the application addressed here because it is the one that results in the highest level of messaging. The amount of messaging involved depends on the complexity of call handling. Simple call handling results in approximately 10 messages per call, with an average of 45 bytes/message. Statistics messages are sent from the system to the CCR module every 4 seconds for ACD DNs referenced in the CCR variable table or scripts. Thus messaging levels depend not only on the number of calls handled but on the number of ACD DNs with statistics configured. Current recommendations are that a system be limited to 80 ACD DNs with statistics.

On the system, messages queue in the CSQI and CSQO buffers, command status queue input and output buffers, which are configurable in LD 17.

Equilibrium analysis

For equilibrium analysis, we focus on calls, and assume ten ACD DNs sending statistics messages. The system capacity assumes an inbound call center with simple CCR treatment on 100% of the calls, and Meridian MAX.

For Large Systems, the CCR module capacity is the system bottleneck (see Table 85: Steady-state requirements and capacities per AML (outgoing and incoming) on page 340). Since there is no flow control or overload control available to protect the CCR module, it is essential that the system be engineered to ensure that the CCR module is not overloaded. Otherwise, link failures or other CCR performance problems may result. To engineer the CCR module, see *Avaya Meridian Link/Customer Controlled Routing Engineering Guide* (553-3211-520).

Table 85: Steady-state requirements and capacities per AML (outgoing and incoming)

System	System CP capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	CCR capacity msg/sec (167 module)	Comment
68060 CP	74	107	41 input 41 output	46	CCR bottleneck
68060E CP	111	107	41 input 41 output	46	CCR bottleneck

Peak analysis

Since message bursts are most likely to cause buffer overflow, we consider the system with 80 ACD DNs sending statistics messages every 4 seconds. Recall that this is the maximum recommended number for ACD DNs sending statistics. The system capacity is based on the real time required to process CCR statistics messages (see Table 86: Peak capacities for CCR statistics messages per AML (outgoing) on page 341).

Table 86: Peak capacities for CCR statistics messages per AML (outgoing)

System	Burst size	System capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	CCR capacity msg/sec (167 module)	Comment
68060 CP	80	920	139	53	60	AML bottleneck
68060E CP	80	1380	139	53	60	AML bottleneck

In this scenario, the AML link is the bottleneck. Messages will begin to queue in the MSDL output buffers and possibly the CSQO buffers, if there are many ACD DNs sending statistics messages.

The AML link will disable if 10 consecutive messages do not receive a response within a 4-second window. The CSA105 message is normally output when this occurs. If a message arrives immediately after the statistics messages for the 80 ACD DNs are generated, it may be queued behind these 80 statistics messages. For 80 messages, processing time at the MSDL, queueing time for the AML, and processing time at the CCR module add up to approximately 3 seconds, so it is easy to understand how the 4 second threshold might be exceeded if the MSDL is also processing messages from other applications.

AML can be configured on the system ELAN network interface. In this configuration, the AML is no longer the bottleneck.

In Meridian Link applications, similar types of problems may occur when the host is too slow and becomes the system bottleneck.

Parameters

On the system side, AML messages are queued in the CSQI/CSQO buffers, which are shared with the CSL. The maximum configurable size of each is 25% of total call registers (NCR). CSQO and CSQI sizes are configured in LD 17.

The flow control parameters MCNT and INTL for each AML are also configured in LD 17. This flow control mechanism limits the number of messages sent from the CCR to the system to MCNT [5.9999] in the time interval INTL [1.12] where INTL is measured in units of 5 seconds. When this threshold is violated for one interval, a warning message is sent to CCR requesting that it slow down. If the threshold is violated for two consecutive periods, CCR rejects all new calls back to the system where they will receive default treatment. No new calls will be accepted until the level of traffic is reduced to an acceptable level. If the threshold is exceeded for three consecutive periods, all inbound traffic will be lost. If inbound traffic continues, the link will fail.

Recommended settings for MCNT and INTL are listed in Table 87: Recommended AML flow control values on page 342.

Table 87: Recommended AML flow control values

MCNT	INTL
230	1

This mechanism was originally designed to protect the system from overload. With the faster processors, this flow control threshold is now being used to control traffic levels at the CCR module.

Serial Data Interface (SDI)

An asynchronous serial data interface was provided on the MSDL card. Capabilities include interface to TTYs, printers, modems, and CRTs, High Speed Link (HSL) for ACD, Auxiliary Processor Link (APL) for ACD, ACD-C package displays and reports, and CDR TTYs. An SDI port is only configurable on Port 0 of an MSDL. Therefore, only one SDI port can be configured on an MSDL.

Normally, in the output direction, the SDI Application will pass 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 will buffer 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 will be 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 will be emptied first, followed by output from the system. If any data has been discarded, an error message will be sent.

In the input direction, every character received by the Layer 1 Driver will be passed to the SDI Application. The SDI Application will echo any input character unless it is told not to by the system. In Line Editing Mode, the SDI Application will buffer a line of up to 80 characters which can be edited before being sent to the system.

Under certain conditions, control characters can cause messages to ping pong between a modem or printer and the system, resulting in MSDL305 or MSDL306 conditions. 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 Records (CDR)

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 88: Link capacities for CDR application (outgoing) on page 343). 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 88: 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

Important:

Throughput capacity for the Quad SDI Paddle Board is the same as the MSDL when operating at the same baud rate. QSDI has a maximum operating baud rate of 9600 bps. Therefore, the maximum throughput for QSDI is 12481 (FCDR=new).

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 (see Table 89: Steady state requirements for CDR application (outgoing) on page 343).

Table 89: Steady state requirements for CDR application (outgoing)

System	System CP capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
68060 CP	2044	16 631	See Table 88: Link capacities for CDR application	19 200 baud recommended

System	System CP capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
			(outgoing) on page 343	
68060E CP	3066	16 631	See Table 88: Link capacities for CDR application (outgoing) on page 343	38 400 baud recommended

Peak analysis

Since each character is sent as a separate message, every time a CDR record is sent, a traffic peak is generated. In Table 90: Peak requirements for FCDR=new (outgoing) on page 344, consider FCDR=new.

Table 90: Peak requirements for FCDR=new (outgoing)

System	Burst size	System capacity (msg/sec)	MSDL capacity (msg/sec)	Link capacity (msg/sec)	Comment
68060 CP	213	3090	21 620	See Table 88: Link capacities for CDR application (outgoing) on page 343	38 400 baud recommended
68060E CP	213	4635	21 620	See Table 88: Link capacities for CDR application (outgoing) on page 343	38 400 baud recommended

MSDL real time capacity is not the bottleneck in this case. However, to prevent system buffers from building up, the recommended baud rate should be set. If a lower baud rate is chosen, assume that the CDR application will frequently be 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 will be even greater if CDR is configured with queue records for incoming ACD calls.

MSDL engineering procedure

It is important to engineer MSDLs in the context of engineering the entire system, as discussed in previous sections. For additional information about real time engineering of the system, see

Avaya Traffic Measurement Formats and Outputs References, (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 these applications messages will occupy system buffers, increasing the chance of buffer overflow.

Table 91: MSDL engineering worksheet on page 345 is the high-level worksheet for analysis of MSDL capacity. The appropriate values can be derived from Table 92: MSDL real time requirements for D-channel applications on page 346 through Table 97: MSDL peak buffer requirements for SDI applications on page 348.

Table 91: MSDL 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 MSDL is given by:

MSDL_RTU = Total Real Time Required ÷ 2 770 000

It is recommended 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 for calculating the real time required on the MSDL for various applications. In any of these cases, 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 should be inserted in the Real Time Required column of Table 91: MSDL engineering worksheet on page 345 and the appropriate Peak Buffer Usage values should be inserted in the corresponding Peak Buffer Usage columns of Table 91: MSDL engineering worksheet on page 345.

DCH applications

If several applications share a D-channel, the final real time requirements for the applications should be added and then entered in the appropriate entry in Table 92: MSDL real time requirements for D-channel applications on page 346.

Table 92: MSDL 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	_____	preR20: 8.8 R20+: 26.5	_____
NACD	NACD agents × calls/hr/ agent (18.3) = _____	30	_____	preR20: 8.8	_____
NMS	NMS ports × calls/hr/ port (65) = _____	10	_____	pre_R20: 8.8	_____
For clarification of the terms "preR20" and "R20+," refer to D-channel on page 330					

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 described in. When this is done, the msgs/hr is computed directly, so columns A and B are not used. See Examples on page 349 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 References, (NN43001-750)* for details.

Table 93: MSDL peak buffer requirements for D-channel applications

DCH	Outgoing	Incoming
ISDN Network	Prior to R24: • B-channels ÷ DCH = _____ R24+: SEMT (1) × 8	Prior to R24: • B-channels ÷ DCH = _____ R24+: SEMT (1) × 8
NACD	Source ACD DNs + 5 = ____	Network congestion level: • Low: 10 • Medium: 20 • High: 30
NMS	10	10

In the case of an ISL D-channel, ensure that the baud rate of the connection is greater than

(C msgs/hr × 29 bytes/msg × 8 bits/byte) ÷ 3600 sec/hr

where C comes from column C in Table 92: MSDL real time requirements for D-channel applications on page 346.

If the baud rate is too low to meet requirements, performance of the entire MSDL card may be jeopardized since 30 of the MSDL output buffers will be occupied with ISL D-channel messages and the real time spent processing these messages will increase due to additional flow control and queueing logic.

Depending on the application, it may be too conservative to engineer an MSDL for link restarts. In that case, the ISDN Network peak outgoing and incoming buffer requirements can be set at 15 for 68060 CP systems.

AML applications

If an existing system is being modeled, add the number of incoming messages, messages in the IMSG category, and outgoing messages, messages in the OMSG category, from a busy hour TFS008 report and enter the value in Column C. For a quick approximation of the number of incoming messages, add the number of messages of priority 1 to 4, as provided in TFS008. For more information, see *Avaya Traffic Measurement Formats and Outputs References, (NN43001-750)*.

Table 94: MSDL real time requirements for AML applications

AML	calls/hr A	msgs/call B	msgs/hr C = A × B	msec/msg D	msec E = C × D
CCR	agents × calls ÷ agent/hr (18.3) × % calls with CCR=_____	simple: 10 medium: 20 complex: 30	A × B + 900 ACD DNs w/ statistics = _____	7.2	____
HER/AST	agents × calls ÷ agent/hr (18.3) × % calls with HER/AST =____	10	_____	7.2	_____
M911	M911 agents × calls ÷ agent/hr (18.3) = ____	6	_____	7.2	_____

Table 95: MSDL peak buffer requirements for AML applications

AML	Outgoing	Incoming	Minimum Baud Rate
CCR	CDNs with statistics= _____	68060 CP: 20 68060E CP:30	(msgs/hr × 45 bytes/msg × 8 bits/byte) ÷ (3600 sec/hr) = ____
HER/AST	68060 CP: 12 68060E CP: 18	68060 CP: 12 68060E CP: 18	(msgs/hr × 45 bytes/msg × 8 bits/byte) ÷ (3600 sec/hr) = ____

AML	Outgoing	Incoming	Minimum Baud Rate
M911	68060 CP: 5 68060E CP: 8	68060 CP: 5 68060E CP:8	(msgs/hr × 45 bytes/msg × 8 bits/byte) ÷ (3600 sec/hr) = ____

SDI applications

In the HSL analysis, include live agents, and automated agents in the agent total. This will compensate for the assumption of simple calls, since transferred calls will generate additional MAX messages.

Table 96: MSDL 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-Meridian MAX	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. MAX reports can also be counted directly.

Table 97: MSDL peak buffer requirements for SDI applications

SDI	Outgoing	Incoming	Minimum baud rate
CDR	• 30 if baud rate is less than recommended in Table 88: Link capacities for CDR application (outgoing) on page 343 • otherwise: - 68060 CP: 20 - 68060E CP:	1	(msgs/hr × 10 bits/msg) ÷ (3600 sec/hr) = ____

SDI	Outgoing	Incoming	Minimum baud rate
HSL – Meridian MAX	• 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 62: NACD network on page 349. 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.

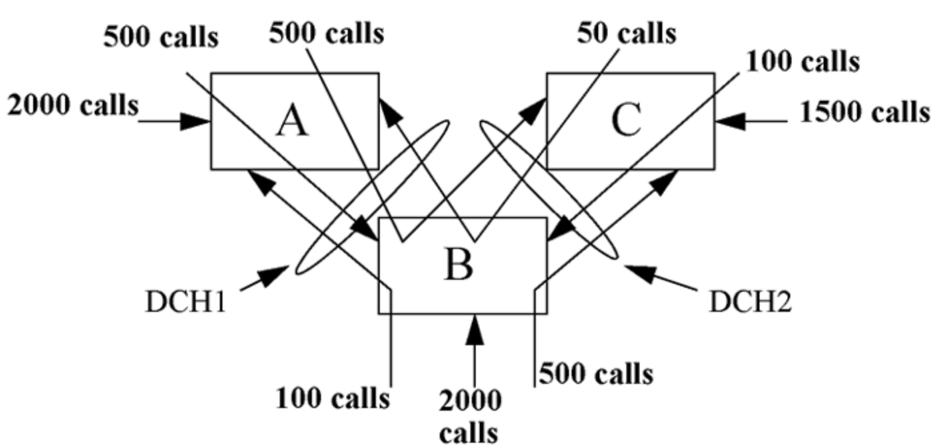

Figure 62: NACD network

For this network, we wish to determine whether a single MSDL 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 98: NACD Message Model on page 350).

Table 98: 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	×	×
Node A to Node C	3000	500	2500	15 500	×	×
Node B to Node A	2600	100	2500	11 100	×	
Node B to Node C	2600	500	2100	13 900		×
Node C to Node A	1650	50	1600	6950	×	×
Node C to Node B	1650	100	1550	7300	×	×

The DCH1 and DCH2 columns indicate whether the messages should 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 which should 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 92: MSDL real time requirements for D-channel applications on page 346).

Table 99: MSDL 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	preR20: 8.8	495 880
NACD DCH2	NA	NA	59 150	preR20: 8.8	520 520

Assuming that no nonNACD calls are carried, Node B carries 3750 calls/hour.

Table 100: MSDL 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 MSDL requirements can then be computed:

Table 101: MSDL 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 MSDL 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 MSDL is able to support this configuration. However, due to the potentially high messaging impact of NACD, this MSDL should be re-engineered periodically to determine whether the call volumes or call flow patterns have changed.

Avaya CallPilot engineering

For more information details, see *CallPilot Planning and Engineering Guide, (555-7101-101)* . The abbreviated procedure in this chapter is for system engineering where a rough estimate of Avaya CallPilot ports (or channels) is required.

The difference in CallPilot engineering is that 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 will reduce the number of channels available for voice traffic.

For an IP source to access CallPilot, the codec must be set for G.711, since a nonstandard proprietary codec is used in CallPilot, a multi-rate transcoding will render 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 will not be covered here. For more information, see *CallPilot Planning and Engineering Guide, (555-7101-101)* .

Once the CCS for each type of media is calculated, sum up the total and refer to capacity tables in the documentation MPU requirement based on the offered CCS traffic. This table was calculated with Erlang C P.05 GoS. An alternative cited in *CallPilot Planning and Engineering Guide, (555-7101-101)*is Erlang B with P.02 GoS, which is slightly on the conservative side compared with the Erlang C model.

Call Center

The Call Center is an ACD switch, whose calls are mostly incoming or outgoing, with extensive applications features, such as CCR, HER, MIVR, HEVP. 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.

Based on customer application requirements, such as calls processed in a busy hour, and feature suite such as RAN, Music, and IVR, the system capacity requirements can be calculated.

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 discussed in this document will have real-time impact on the system CP capacity.

ACD-C1 and C2 packages

ACD Management Reporting provides the ACD customer with timely and accurate statistics relevant to the ACD operation. These statistics form periodic printed reports and ongoing status displays so the customer can monitor changing ACD traffic loads and levels of service and implement corrective action where required.

The ACD-C1 package primarily provides status reporting of the system through a TTY terminal. To control and alter the configuration of the system, the ACD-C2 package is required; it provides the load management commands. The following is a partial list of functions of a supervisor position in the C2 package:

- Assign auto-terminating ACD trunk routes
- Assign priority status to ACD trunks
- Reassign ACD agent positions to other ACD DNs
- Set 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.

The impact of ACD-MAX calls in the capacity engineering will be in the real-time area only. The Meridian MAX is an AP version of the ACD-MAX which uses an AP module instead of an HP computer as an auxiliary processor. To estimate the impact of MAX on the system CP, both versions can be treated the same.

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.

Meridian 911

The primary difference between the M911 application and other Application Module link related incoming ACD calls is the requirement of MF Receivers (MFR), which interpret digits received from CO through MF trunks for M911 calls.

Estimating MFR requirements

1. Calculate the number of calls from MF trunks:

   M911 calls = No. of MF trunks × 28 × 100 ÷ 180 = 15.56 × No. of MF trunks.

   where the default value of CCS for the trunk is 28 and the average holding time is 180 seconds. These numbers should be replaced by specific values at your site if they are available.
2. Calculate MFR traffic:

   MFR traffic in CCS = M911 calls × 6 ÷ 100

   where the ANI digits of 8 were estimated conservatively to hold up a receiver for 6 seconds.
3. To find the requirements of MFRs, see *Avaya Dialing Plans Reference, (NN43001-283)* . For the purpose of estimating MFR requirements, the DTR table can be applied. Read the number of DTRs (MFRs) corresponding to a CCS entry greater than the above calculated CCS value under the column of 6-second holding time. An abbreviated table is shown here for simple reference.

   Table 102: MFR table with 6-second holding time

No. of MF receivers	2	4	6	8	10	15	20	25	30	35	40
Capacity in CCS	3	24	61	106	157	300	454	615	779	947	1117

RAN and Music

The RAN trunk can be treated just like a normal trunk. The only potential capacity impact is for Large Systems that include RAN trunks in blocking or nonblocking calculations. The calculations determine the total number of loops or card slots required. To calculate RAN requirements, see Service loops and circuits on page 213.

Music in the system is provided by broadcasting a music source from a RAN trunk to a conference loop. Therefore, a maximum of 30 users can listen to music at one time. If this is not sufficient, an additional conference loop needs to be provided for each additional 30 simultaneous music users.

The conference loop connects to one half of the Conference/TDS card. The second conference loop, if needed, will take another card and card slot, because it cannot be separated from the TDS loop.

Music Broadcast requires any Music trunk and an external music source or a Meridian Integrated RAN (MIRAN) card (NTAG36). MIRAN has the capability to provide audio input for external music. A Conference loop is not required for Music Broadcast.

Symposium Call Center

Symposium is a Host Server that interfaces through an Ethernet to reach Meridian's Network Interface Card to enable the system to provide advanced Call Center features to users. The NIC port can be set for many options, such as 10T (10 Mbps), or 100T (100 Mbps) data rate, half duplex or full duplex depending on processor capacity of the system. For CP PIV, all options are available, including full duplex and 100T data rate. Although Internet Protocol (IP) is used for communications, the underlining message to Meridian input queue is still AML messages.

The customer can create simple to write script in Symposium to control processing of an arriving call which is eventually delivered to an agent queue after following various call processing rules, such as skill set of agent, call priority, length of waiting time, etc.

The impact of Symposium call center on the system is the complexity of call handling on the system call processor. Depending on the script used, the call processing can include giving RAN, Music and IVR which requires a voice processing system such as Avaya CallPilot be also provided.

From the system CPU's point of view, service functions provided by Symposium are similar to that available for CCR, or HEVP, the only difference is reflected by real time factor corresponding to each application type.

Symposium Call Center with IP phones and Virtual Trunks

When IP phones are used as ACD agent telephones, some special engineering rules are to be followed to engineer the system properly. Two new resources must be engineered:

- Digital Signaling Processing (DSP) channels (therefore, Media Cards)
- Virtual Trunks

The following four configurations demonstrate the application of different rules to accommodate different configurations.

1. New Pure IP System with IP telephones and VTs

   In a pure IP system, if Signaling Processing and Gateway functions are provided, no DSP channels are needed for pure IP connections. The number of VTs provided must be equal to or more than the number of IP agent telephones depending on the queuing provisioning. Typically, trunks should exceed agents by 15-20%.

2. PRI trunks and IP agent telephones

   One DSP channel per IP agent telephone. How many more PRI trunks than IP agent telephones depends on the queueing consideration.

3. Mixed PRI trunks and Virtual Trunks to IP agent telephones

   For DSP channel calculation, consider only the number of PRI trunks and required IP agent telephones. The subset of VTs versus IP agents can be excluded. However, for bandwidth calculation, all traffic must be accounted for.

4. Mixed TDM and IP Call Center

   When both PRI trunks and VTs carry traffic to agents with digital telephones and IP telephones, the first step is to determine whether there is a community of interests among PRI trunks and digital agent telephones. If so, their connections are preferred through the control of CDN, making codecs unnecessary in the call set up which reduces usage of DSP resources and maintains high QOS.

ELAN subnet engineering

The 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 the function of the enterprise IP network which carries customer application traffic.

A 64 kbps link can handle messaging traffic of over 80 000 calls. The ELAN subnet, being an Ethernet subnet with a data rate of 10/100/1000 Mbps, is not a bottleneck in a Symposium/

CallPilot configuration. However, certain engineering guidelines must be followed to avoid any performance problems:

- Settings on the Network Interface Card (NIC), the physical interface of the system to the ELAN subnet, must be properly configured in order to guarantee smooth operation. The CP PIV can handle half duplex or full duplex and 10/100/1000 Mbps data rate.
- It is important to set a consistent data rate for the NIC and application.

Certain remote maintenance applications, including Element Manager may use the ELAN subnet to access the system from a remote location. Ensure no other enterprise IP network traffic is introduced.

CLASS network engineering

In a single-group network system, the network internal blocking is determined by the concentration ratio of equipped ports on 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 B starts to ring, A hears ringback. A unit in CLASS Modem (CMOD) is assigned to collect originator's CND information and waits for the CND delivery interval. After the first ring at 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 (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, A's CND is delivered to CMOD through SSD messages (using the 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.

If the Extended CLASS Modem Card (XCMC) IPE card, which provides up to 32 CMOD units, is located in the IPE of Group 0, the CMOD unit in the card receives CND data through the SSD messages and uses one of the voice channels of the intergroup junctor to deliver it to CLASS Telephone B in Group 1.

If the XCMC IPE card is located in Group 1, the system delivers SSD messages containing CND information to CMOD and then send it to Telephone B during the delivery interval through a voice path, which is an intra-group channel not involving an intergroup junctor.

When CMOD units and CLASS telephones are collocated in the same network group, there are no voice paths on the intergroup junctor required to deliver CND information; when they are equipped on different groups, intergroup junctors must carry CND traffic. The resource allocation algorithm searches for a CMOD unit located in the same group as the terminating CLASS telephone first before it attempts to use a CMOD unit from a different group.

The process continues to be valid. However, due to nonblocking on the fiber link, a multi-group system can be treated as a single group system, since intergroup blocking no longer exists.

Fiber Network Fabric

Multi-group networks are interconnected with fiber-optic rings. The OC-12 fabric has such a large capacity that all channels from expanded eight network groups can be interconnected without junctor blocking. Therefore, engineering of the CLASS feature is reduced to the equivalent of a single group case. The only engineering required is to find the required number of CMOD units from Table 103: CMOD Unit Capacity on page 358 to serve a given number of CLASS telephones. Capacity limit due to network group size can be ignored.

Table 103: CMOD Unit Capacity on page 358 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 must have a capacity range with an upper limit that is greater than the number of CLASS telephones equipped in a given configuration.

Table 103: 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

CMOD Unit	CLASS Telephone	CMOD Unit	CLASS Telephone
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
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 agent telephone to regular telephone conversion is required.

Configurations following engineering rule (no reconfiguration required)

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 (see Table 102).

Engineering example

One XCMC card serving a single-group system

No special engineering rule is needed for a single-group system.

To find the required number of CMOD units to serve the given CLASS telephones, see Table 103: CMOD Unit Capacity on page 358. For example, to serve a Meridian 1 Option 61Cwith 400 CLASS telephones, use Table 103: CMOD Unit Capacity on page 358 to find the number of CMOD units serving a range that includes 400 telephones. The result is 10 units, which can serve from 333 - 401 CLASS telephones.

Guidelines for Call Center applications

Configurations following engineering rules (no reconfiguration required)

The following engineering rules should be followed 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. Calculate 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 103: CMOD Unit Capacity on page 358).

Engineering example

One XCMC card serving a single-group system

For the number of CMOD units required to serve the given CLASS telephones, see Table 103: CMOD Unit Capacity on page 358. For example, to serve a Meridian 1 Option 61Cwith 300 agent CLASS telephones, use Table 103: CMOD Unit Capacity on page 358 to find the CMOD units that can serve 1200 equivalent telephones (300 × 4). The result is 20 units.

Configuration parameters

Design parameters are constraints on the system established by design decisions and enforced by software checks. A complete list is provided in the Appendix, with default values, maximums and minimums, where applicable. Although defaults are provided in the factory installed database, the value of some of these parameters are configured 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 Mass storage on page 197.

Chapter 16: Provisioning

Contents

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Calculations with Direct Inward System Access on page 375

Introduction

To determine general equipment requirements, follow the provisioning steps in the order shown below. (These provisioning methods are based on a nonpartitioned system.) Use the worksheets prepared in the previous chapter and the reference tables at the end of this document.

Step 1: Define and forecast growth on page 364

Step 2: Estimate CCS per terminal on page 365

Step 3: Calculate number of trunks required on page 369

Step 4: Calculate line, trunk, and console load on page 370

Step 5: Calculate Digitone receiver requirements on page 371

Step 6: Calculate total system load on page 376

Step 7: Calculate number of network loops required on page 376

Step 8: Calculate number of network groups required on page 377

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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. Consult your Avaya representative and use a configuration tool, such as Enterprise Configurator, to fully engineer a system.

Step 1: Define and forecast growth

The first step in provisioning a new system is to forecast the number of telephones required at two-year and five-year intervals.

The number of telephones required when the system is placed in service (cutover) is determined by the customer. If the customer is unable to provide a two-year and five-year growth forecast, then an estimate of annual personnel growth in percent is used 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, the number of employees required at cutover, one, two, three, four, and five years is multiplied by the ratio of telephones to employee (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 and enter this information in Worksheet 13:Network loop balancing on page 395. 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) for the installation of a system using any one of the following methods:

1. Comparative method
2. Manual calculation
3. Default method

Comparative method

Select three existing systems that have a 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)

Once similar systems have been selected, then their station, trunk, and intra-CCS/T are averaged. The averages are applied to calculate trunk requirements for the system being provisioned (see the example in Table 104: Example of station, trunk, and intra-CCS/T averaging on page 366).

Table 104: Example of station, trunk, and intra-CCS/T averaging

	Customer A	Customer B	Customer C	Total	Average
Line size	200	250	150	600	200
Line CCS/T	4.35	4.75	3.50	12.60	4.20
Trunk CCS/T	2.60	3.00	2.00	7.60	2.53
Intra CCS/T	1.70	1.75	1.50	4.95	1.65

If only the trunk CCS/T is available, multiply the trunk CCS/T by 0.5 to determine the intra-CCS/T (assuming a normal traffic pattern of 33% incoming calls, 33% outgoing calls, and 33% intra-system calls). The trunk CCS/T and intra-CCS/T are then added to arrive at the line CCS/T (see the example in Table 105: Example of CCS/T averaging when only trunk CCS/T is known on page 366).

Table 105: Example of CCS/T averaging when only trunk CCS/T is known

Trunk type	Number of trunks	Grade-of-Service	Load in CCS	Number of terms	CCS/T
IP Peer Virtual Trunk	11	P.01	169	275	0.61
DID	16	P.01	294	234	1.20
CO	14	P.02	267	234	1.14

Trunk type	Number of trunks	Grade-of-Service	Load in CCS	Number of terms	CCS/T
TIE	7	P.05	118	215	0.54
Paging	2	10 CCS/trunk	20	207	0.09
Out WATS	4	30 CCS/trunk	120	218	0.54
FX	2	30 CCS/trunk	60	218	0.27
Private line	4	20 CCS/trunk	80	275	0.29
			Total: 1128		Total: 4.64
The individual CCS/T per trunk group is not added to form the trunk CCS/T. The trunk CCS/T is the total trunk load divided by the total number of lines at cutover.					

From the preceding information, trunk CCS/T can be computed as follows:

Trunk CCS/T = total trunk load in CCS ÷ (number of lines) = 1128 ÷ 275 = 4.1

Assuming a 33% intra-calling ratio:

Intra CCS/T = 4.1 × 0.5 = 2.1, and line CCS/T = 4.1 (trunk CCS/T) + 2.1 (intra-CCS/T) = 6.2

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 106: Example of manual calculation of CCS/T on page 367). (If not, use the comparative method.)

The number of trunks can be read from the appropriate trunking table to select the estimated usage on the trunk group. The number of lines that are accessing the group at cutover are divided into the estimated usage. The result is the CCS/T, which can be used to estimate trunk requirements.

Example

- Line CCS/T = 6.2
- Trunk CCS/T = 4.1
- 2 consoles = 30 CCS

Table 106: Example of manual calculation of CCS/T

Cutover	Line CCS = 275 × 6.2 = 1705

	Trunk CCS = 275 × 4.1 =1128 Subtotal = 2833 Console CCS = 30
	Total system load = 2863
2 years	Line CCS = 304 × 6.2 = 1885 Trunk CCS = 304 × 4.1 = 1247 Subtotal = 3132 Console CCS = 30
	Total system load = 3162
5 years	Line CCS = 352 × 6.2 = 2183 Trunk CCS = 352 × 4.1 = 1444 Subtotal = 3627 Console CCS = 30
	Total system load = 3657

This method is used for each trunk group in the system, with the exception of small special services trunk groups (such as TIE, WATS, and FX trunks). Normally, the customer will tolerate a lesser GoS on these trunk groups. Table 107: Estimated load per trunk on page 368 lists the estimated usage on special services trunks.

Table 107: Estimated load per trunk

Trunk type	CCS
IP Peer Virtual Trunk	30
TIE	30
Foreign exchange	30
Out WATS	30
In WATS	30
Paging	10
Dial dictation	10
Individual bus lines	20

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.

The network line usage is determined by multiplying the number of lines by 5.5 CCS/T. The total is then multiplied by 2 to incorporate the trunk CCS/T. However, when this method is used, the intra-CCS/T is added twice to the equation, and the result could be over provisioning if the intra-CCS/T is high.

Another difficulty experienced with this method is the inability to forecast individual trunk groups. The trunk and intra CCS/T are forecast as a sum group total. Examples of the default method and the manual calculation method are shown in Table 108: Default method and manual calculations analysis on page 369 for comparison.

Example

- 275 stations at cutover
- 304 stations at two years
- 352 stations at five years

Cutover	275 × 5.5 (CCS/T) × 2 =	3025 CCS total system load
Two-year	304 × 5.5 (CCS/T) × 2 =	3344 CCS total system load
Five-year	352 × 5.5 (CCS/T) × 2 =	3872 CCS total system load

Table 108: Default method and manual calculations analysis

	Default method	Manual calculations	Difference
Cutover	3025	2863 CCS	162 CCS
Two years	3344	3162 CCS	182 CCS
Five years	3872	3657 CCS	215 CCS

Step 3: Calculate number of trunks required

Enter the values obtained through any of the three previous methods in Worksheet 10:Growth forecast on page 392. Add the calculations to the worksheet. Once the trunk CCS/T is known and a GoS has been specified by the customer, the number of trunks required per trunk group to meet cutover, two-year, and five-year requirements is determined as shown in the following example.

Example

The customer requires a Poisson 1% blocking GoS (see Trunk traffic Erlang B with P.01 Grade-of-Service on page 409). The estimated trunk CCS/T is 1.14 for a DID trunk group. With the cutover, two-year, and five-year number of lines, the total trunk CCS is determined by multiplying the number of lines by the trunk CCS/T:

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 Digitone receiver requirements Model 2 on page 414 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

Once the quantity of trunks required has been estimated, enter the quantities in Worksheet 10:Growth forecast on page 392 for cutover, two-year, and five-year intervals. This calculation must be performed for each trunk group to be equipped. The total trunk CCS/T is the sum of each individual trunk group CCS/T. This value is also entered inWorksheet 10:Growth forecast on page 392.

Line load

Line load is calculated 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

Trunk load is calculated by multiplying the total number of single- and multi-line TNs that have access to the trunk route by the CCS/T per trunk route.

Console load

Console load is calculated 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, the DTR requirements can be calculated. The DTRs are shared by all customers in the system and must be distributed equally over all the network loops.

The tables Digitone receiver requirements Model 3 on page 415 through Digitone receiver load capacity 16 to 25 second holding time on page 419 are based on models of traffic environments and can be applied to determine DTR needs in most cases. When the system being provisioned does not fall within the bounds of these models or is equipped with any special features, the detailed calculations must be performed for each feature and the number of DTRs must accommodate the highest result.

Special feature calculations include:

- Calculations with Authorization Code on page 374
- Calculations with Centralized Attendant Service on page 374
- Calculations with Charge Account for Call Detail Recording on page 375
- Calculations with Direct Inward System Access on page 375

From the appropriate reference table (Digitone receiver requirements Model 1 on page 414 through to Digitone receiver load capacity 16 to 25 second holding time on page 419), determine the number of DTRs required and the DTR load for cutover, two-year, and five-year intervals. Record this information in Worksheet 11:Total load on page 393.

The following models are based on some common PBX traffic measurements.

Model 1

Digitone receiver requirements Model 3 on page 415 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 4 on page 416 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 load capacity 6 to 15 second holding time on page 417 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 load capacity 16 to 25 second holding time on page 419 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

Detailed calculation Method 1

This method can be used when there are no incoming Digitone DID trunks and the following is assumed:

- Digitone receiver traffic is inflated by 30% to cover unsuccessful dialing attempts.
- Call holding time used in intraoffice and outgoing call calculations is 135 seconds if unknown.

- Digitone receiver holding times are 6.2 and 14.1 seconds for intraoffice and outgoing calls, respectively.
- Factor (1 – R) ÷ 2 in (1) outgoing (incoming calls and outgoing calls are equal). R is the intraoffice ratio.

Follow Detailed calculation Method 1 on page 373 to complete a detailed calculation using Method 1.

Detailed calculation Method 1

1. Calculate Digitone calls:

   Intraoffice = 100 × Digitone station traffic (CCS) ÷ call holding time × (R ÷ 2)

   Outgoing = 100 × Digitone station traffic (CCS) ÷ call holding time × [(1 – R) ÷ 2]
2. Calculate total DTR traffic:

   1.3 × [(6.2 × Intra) + (14.1 × Outgoing)] ÷100
3. Calculate average holding time:

   (6.2 × intra) + (14.1 × outgoing) × intra calls + outgoing calls
4. See Digitone receiver requirements Poisson 0.1 percent blocking on page 420 or Conference and TDS loop requirements on page 421 and use the answers from steps 2 and 3 to determine the number of DTRs required.

Detailed calculation Method 2

This method is used when incoming Digitone trunks are included in the system. This method uses the same assumptions as Method 1, with the DTR holding time assumed to be 2.5 seconds for a DID call.

Follow Detailed calculation Method 1 on page 373 to complete a detailed calculation using Method 2.

Detailed calculation Method 1

1. Calculate intraoffice and outgoing Digitone calls as shown in step 1 of Method 1:

   DID calls = DID Digitone trunk traffic (CCS) × 100 ÷ call holding time
2. Calculate total DTR traffic:

   [(1.3 × 6.2 × intra) + (1.3 × 14.1 × outgoing calls) + (2.5 × DID calls)] ÷ 100
3. See Table 128: Digitone receiver requirements Poisson 0.1 percent blocking on page 420 and use the answer from step 2 to determine the number of DTRs required.

Calculations with Authorization Code

With Authorization Code, the DTR holding times change from 6.2 seconds to 19.6 seconds for intraoffice calls, and from 14.1 seconds to 27.5 seconds for outgoing calls.

Use the values in steps 2 and 3 of Detailed calculation Method 1 on page 372 and step 2 of Detailed calculation Method 2 on page 373 to calculate the DTR requirements for a system with the Authorization Code option.

It has been assumed that:

1. all Digitone intraoffice and outgoing calls require authorization;
2. the average number of special services prefix (SPRE) digits is two (the maximum is four);
3. the average number of Authorization Code digits is 10 (the range is 1 to 14 digits); and,
4. the average DTR holding time is 13.4 seconds.

Calculations with Centralized Attendant Service

This method determines the DTR requirements for the main location of a system equipped with the CAS option. It has been assumed that:

1. all attendant calls presented through release link trunks from a remote PBX require DTRs;
2. the average number of digits dialed is four; and,
3. the average DTR holding time is 6.2 seconds.

Determining DTR requirements

1. Calculate the attendant calls from the remote PBX: 100 × attendant traffic from the remote (CCS) ÷ attendant work time (in seconds)
2. Add the attendant calls to the intraoffice calls calculated in step 1 of "Detailed calculation: Method 1" and proceed with the remaining calculations of Method 1.

Calculations with Charge Account for Call Detail Recording

The DTR holding time for outgoing calls changes from 14.1 seconds to 20.8 seconds.

Apply this change to steps 2 and 3 of Detailed calculation Method 1 on page 372 and step 3 of Detailed calculation Method 2 on page 373 to determine the DTR requirements for a system with the Charge Account for CDR option.

It has been assumed that:

1. 50% of Digitone outgoing calls require a charge account;
2. the average number of SPRE digits is two (maximum is four);
3. the average number of digits in the account number is 10 (the range is 2 to 23 digits); and,
4. the average DTR holding time is 13.4 seconds (see Digitone receiver requirements Poisson 0.1 percent blocking on page 420).

Calculations with Direct Inward System Access

This method is used when a system is equipped with the DISA feature. It has been assumed that:

1. DISA calls come through DISA trunks or DID trunks;
2. 75% of DISA calls require a security code;
3. the average number of digits in the security code is four (the range is one to eight); and,
4. the DISA DTR holding time is 6.2 seconds.

Determining DTR requirements

1. Calculate the number of DISA calls: 100 × DISA traffic ÷ call holding time
2. Calculate the DISA DTR traffic: 6.2 × DISA calls ÷ 100
3. Add this traffic to step 2 of "Detailed calculation: Method 2" and proceed with the remaining calculations of Method 2.

Step 6: Calculate total system load

Total the line, trunk, console, and DTR load for each customer to get the total load figure for each customer for cutover, two-year, and five-year intervals. Enter this figure in Worksheet 11:Total load on page 393 and Worksheet 12:Network loops on page 394.

Step 7: Calculate number of network loops required

The system network loop requirement is the total of all individual customer loops and superloops required. The number of network loops and superloops required is calculated for each customer for cutover, two-year, and five-year intervals. Network loops and superloops are provisioned at cutover based on the two-year loop requirement figure.

To determine the number of superloops required, first separate the traffic supported by data line cards, RPE, and PRI/DTI. The remaining traffic (including DTR traffic) must be engineered for superloops.

Number of superloop network cards or number of superloops = traffic to be handled by superloop network ÷ 2975

These figures are based on an 85% utilization level. Round the value obtained to the next higher number.

Nonblocking configuration with superloop network

For nonblocking applications (or a nonblocking part of the system), provide one superloop for every 120 TNs. Generally, each line or trunk is one TN, but an integrated voice and data line is two TNs (assuming the data port is configured). Application processors such as Avaya CallPilot and MICB require a TN for each port. Media Card requires 1 TN for each DSP port.

Blocking configuration with superloop network

For applications where blocking is allowed, one superloop can serve up to 512 lines (1024 TNs). The actual number of lines depends on the traffic requirement of the lines.

PRI/DTI cards

The PRI and DTI cards provide the interface between the system switch and T-1/DS-1 digital transmission trunks. Digital trunks are offered in a group of 24 trunks. Table 109: Number of cards required when PRI/DTI traffic is known on page 377 lists the number of PRI/DTI cards required when PRI/DTI traffic is known.

The Line-side E1 Interface card (LEI) is an IPE line card that provides a cost-effective, all-digital connection between E1 compatible terminal equipment (such as voice mail systems, voice response units, trading turrets, etc.) and the system. In this application it will provide 30 ports.

The number of PRI/DTI loops is the same as the number of PRI/DTI cards.

Table 109: Number of cards required when PRI/DTI traffic is known

Number of cards	CCS for 24-port T1	CCS for 30-port E1
1	1-507	1-675
2	508-1201	676-1565
3	1202-1935	1566-2499
4	1936-2689	2500-3456
5	2690-3456	3457-4427
6	3457-4231	4428-5409
7	4232-5015	5410-6403
8	5016-5804	6404-7428

For nonblocking applications, the Ring Again feature must be provided since blocking may occur at the far end of the trunk.

The PRI/DTI cards can be installed in any module except IPE Modules. After all essential cards are configured, estimate the available slots for PRI/DTI.

Step 8: Calculate number of network groups required

Compute the number of network groups based on the total number of loops required (excluding conference/TDS loops). Record the network groups in Worksheet 13:Network loop balancing on page 395. Use Table 110: Number of network groups based on total number of loops required on page 378 and the following equation to find the number of network groups required:

Total number of loops = (4 × the number of superloop network cards)

Table 110: Number of network groups based on total number of loops required

Number of network groups	Number of loops
1	28
2	56
3	84
4	112
5	140
6	168
7	196
8	224
Use Worksheet 12:Network loops on page 394. Install a multiple-group system if the total number of loops required exceeds 28.	

Based on the criteria above, installing a multiple group system initially is more cost-effective than converting to a multiple group system (from a single-group system) between the two-year and five-year intervals.

For Avaya Communication Server 1000M (Avaya CS 1000M) MG and Meridian 1 Option 81C CP PIV use Table 111: cCNI configurations (Avaya CS 1000M) MG and Meridian 1 Option 81C CP PIV) on page 378 to calculate the number of NTRB34 cCNI Cards needed to support the network groups.

Table 111: cCNI configurations (Avaya CS 1000M) MG and Meridian 1 Option 81C CP PIV)

Number of network groups supported	Required number of cCNI cards
1 (group 0)	1
2 (group 1)	1
3 (group 2)	2
4 (group 3)	2
5 (group 4)	3
6 (group 5)	3
7 (group 6)	4
8 (group 7)	4

Step 9: Calculate number of IPE cards required

InWorksheet 14:IPE card calculations on page 396, enter the number of DTRs required (fromWorksheet 11:Total load on page 393). Use a separate worksheet for cutover, two-year, and five-year intervals.

Using information from Worksheet 10:Growth forecast on page 392, enter the number of single-line telephone TNs, multiline telephone TNs, and trunk TNs required at cutover, two-year, and five-year intervals (for all customers) in Worksheet 14:IPE card calculations on page 396.

Divide each entry by the number of TN assignments for each card, round up to the next higher figure, and total the number of cards required.

Calculate the number of IPE cards separately.

Step 10: Calculate number of IPE modules required

The number of Peripheral Equipment modules provided at cutover is based on the two-year estimate of Peripheral Equipment cards required and an 85% utilization level.

The maximum capacity of an IPE Module is 256 integrated voice and data or analog lines; however, a typical configuration includes a combination of lines, trunks, and DTRs, which provides up to 160 lines.

Divide the number of Peripheral Equipment cards required at two years by 8.5, round to the next higher number, and enter this value in Worksheet 14:IPE card calculations on page 396.

To compute the number of Peripheral Equipment modules, divide the total number of line, trunk, and DTR cards required at two years by 13.6 and round to the next higher number. Enter this value in Worksheet 14:IPE card calculations on page 396.

Calculate the number of IPE Modules required.

Step 11: Provision conference/TDS loops

Conference/TDS loops are provisioned according to the two-year figure for the number of network loops required. All systems must be equipped with a minimum of two conference and two TDS loops.

See Table 108: Default method and manual calculations analysis on page 369 and Table 109: Number of cards required when PRI/DTI traffic is known on page 377 to determine conference/ TDS loop requirements. Enter these figures in Worksheet 15:Conference and TDS loop requirements on page 397.

Step 12: Assign equipment and prepare equipment summary

Use Worksheet 17: Equipment summary on page 402 to record the equipment requirements for the complete system at cutover. Assign the equipment. The equipment summary may have to be updated as a result of assignment procedures. Use the finalized equipment summary to order the equipment for the system.

Appendix A: Worksheets

List of worksheets

Worksheet 1:Load balancing on page 383

Worksheet 2:Circuit card distribution on page 384

Worksheet 3:Multiple appearance group assignments on page 385

Worksheet 4:Station load balancing on page 386

Worksheet 5:Multiple appearance group record on page 387

Worksheet 6:Circuit card to module assignment on page 388

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Worksheet 13:Network loop balancing on page 395

Worksheet 14:IPE card calculations on page 396

Worksheet 15:Conference and TDS loop requirements on page 397

Worksheet 16: Resource calculations on page 397

Worksheet 17: Equipment summary on page 402

Worksheet 18: Network loop traffic capacity on page 402

Worksheet 19: Physical capacity on page 403

Worksheet 20: Signaling Server calculation on page 404

Worksheet 21: Real time calculation on page 405

Introduction

The worksheets in this appendix provide examples of information required to calculate power consumption, allocate circuit cards, and do traffic and equipment engineering. However, more detailed information is required to fully engineer a system. Consult your Avaya representative and use a configuration tool, such as EC, to fully engineer a system.

Each traffic and equipment engineering subsection contains a worksheet with which the system engineer can assess the total system impact of a given configuration on the specified capacity. These worksheets implement the algorithms described in Resource calculations on page 247. The result of the worksheet is a number or set of numbers, in the units of the capacity being assessed. A simplified table of capacity limits is given to provide easy determination of feasibility and the size of system required.

Worksheet 1:Load balancing

LOAD BALANCING

CUSTOMER ________________________________ DATE __________

One sheet for the complete system.

Total system load = ________ CCS

Voice loops required = ________

IPE/PE modules required = ________

Average CCS per module = $\frac{\text{Total system load CCS}}{\text{IPE modules required}}$ = ________ CCS

Average CCS per loop = $\frac{\text{Total system load CCS}}{\text{Voice loops required}}$ = ________ CCS

LOOP NUMBER	SHELVES ASSIGNED	CCS PER LOOP	CCS PER SHELF

553-5366

Worksheet 2:Circuit card distribution

CIRCUIT CARD DISTRIBUTION

CUSTOMER ______________________ DATE __________

One sheet for the complete system.

Divide the total number of a card type by the total number of IPE modules to arrive at a cards-per-module number.

CARD TYPE	QUANTITY	TOTAL IPE MODULES	CARDS PER MODULE

553-5367

Worksheet 3:Multiple appearance group assignments

MULTIPLE APPEARANCE GROUP (MAG) ASSIGNMENTS

CUSTOMER ______________________________ DATE __________

One sheet for the complete system.

LOOP #	LOOP #	LOOP #	LOOP #	LOOP #
MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN
MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN
MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN
MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN
MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN	MAG # Single-line TN Multi-line TN
CARDS Single-line ____ Multi-line ____	CARDS Single-line ____ Multi-line ____	CARDS Single-line ____ Multi-line ____	CARDS Single-line ____ Multi-line ____	CARDS Single-line ____ Multi-line ____

553-4054

Worksheet 4:Station load balancing

STATION LOAD BALANCING
CUSTOMER________________________________ DATE __________

One sheet required for the complete system.

Total single-line TNs to be assigned		______________
Less number of single-line TNs assigned to MAG	–	______________
Equals number of single-line TNs not in MAG	=	______________

$$\frac{\text{Single-line TNs not in MAG}}{\text{Total IPE modules}} = \_\_\_\_\_\_\_\_\_\_ \quad \begin{array}{l}\text{Number of single-line TNs not in MAG} \\ \text{Assigned per module}\end{array}$$

Total multi-line TNs to be assigned		______________
Less number of multi-line TNs assigned to MAG	–	______________
Equals number of multi-line TNs not in MAG	=	______________

$$\frac{\text{Multi-line TNs not in MAG}}{\text{Total IPE modules}} = \_\_\_\_\_\_\_\_\_\_ \quad \begin{array}{l}\text{Number of multi-line TNs not in MAG} \\ \text{Assigned per module}\end{array}$$

553-5372

Worksheet 5:Multiple appearance group record

MULTIPLE APPEARANCE GROUP (MAG) RECORD

CUSTOMER ______________________ DATE __________

One worksheet per system.

MAG NUMBER	LOOP NUMBER	DN	STATUS DESIGNATOR	TELEPHONE TYPE	DEPT.

553-4059

Worksheet 6:Circuit card to module assignment

CIRCUIT CARD TO MODULE ASSIGNMENT

CUSTOMER________________________________ DATE __________

One table for each IPE shelf in the system.

LOOP # ________ MODULE # ________

Position	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	TOTAL CARDS	CCS LOAD
Type																	

LOOP # ________ MODULE # ________

Position	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	TOTAL CARDS	CCS LOAD
Type																	

LOOP # ________ MODULE # ________

Position	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	TOTAL CARDS	CCS LOAD
Type																	

LOOP # ________ MODULE # ________

Position	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	TOTAL CARDS	CCS LOAD
Type																	

LOOP # ________ MODULE # ________

Position	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	TOTAL CARDS	CCS LOAD
Type																	

553-5369a

Worksheet 7:Terminal number assignment

TN ASSIGNMENT RECORD

CUSTOMER ____________________ DATE __________

LOOP # ________ MODULE # ________ GROUP # ________

CARD POS __________ CARD POS __________ CARD POS __________

CARD TYPE __________ CARD TYPE __________ CARD TYPE __________

UNIT	DN	RTMB	CUST	UNIT	DN	RTMB	CUST	UNIT	DN	RTMB	CUST
0				0				0			
1				1				1			
2				2				2			
3				3				3			
4				4				4			
5				5				5			
6				6				6			
7				7				7			
8				8				8			
9				9				9			
10				10				10			
11				11				11			
12				12				12			
13				13				13			
14				14				14			
15				15				15			
16				16				16			
17				17				17			
18				18				18			
19				19				19			
20				20				20			
21				21				21			
22				22				22			
23				23				23			
24				24				24			
25				25				25			
26				26				26			
27				27				27			
28				28				28			
29				29				29			
30				30				30			
31				31				31			

DN = Directory Number, RTMB = Route Member Number (trunks)

553-5368

Worksheet 8:System assignment plan

SYSTEM ASSIGNMENT PLAN

CUSTOMER____________________________ DATE __________

One sheet for each equipment voice loop.

LOOP #: __________ GROUP #: __________

Modules equipped	__________
Trunks working	__________
Trunks equipped	__________
Consoles	__________
DTRs	__________
Single-line TNs	__________
Multi-line TNs	__________
MAGs assigned	__________
Load capacity	__________

RECOMMENDED ASSIGNMENT PLAN ____________________

553-5370

Worksheet 9:System power consumption

SYSTEM POWER CONSUMPTION WORKSHEET

Module	Quantity		Module power consumption		Total module power consumption
NT4N41	________	X	53.5	=	________
NT5D21	________	X	260	=	________
NT6D44	________	X	240		________
NT6D60	________	X	260	=	________
NT8D35	________	X	240	=	________
NT8D37	________	X	460	=	________
Pedestals	________	X	50	=	________
			Total real power (watts)	=	________

Current drain

AC system:

$$\frac{\text{(total real power)}}{\text{(nominal AC voltage)} \quad 208} = \_\_\_\_\_\_\_\_ \text{ amperes, AC}$$

DC system:

$$\frac{\text{(total real power)}}{\text{(nominal AC voltage)} \quad 52} = \_\_\_\_\_\_\_\_ \text{ amperes, DC}$$

Complex (or apparent power) (AC only):

$$\frac{\text{(total real power)}}{\text{(power factor)} \quad 0.6} = \_\_\_\_\_\_\_\_ \text{ volt-amperes (VA)}$$

553-AAA1661

Worksheet 10:Growth forecast

GROWTH FORECAST

CUSTOMER ______________________________ DATE __________

One sheet for each customer and one sheet for the system as a whole.

	CUTOVER	2-YR	5-YR	CCS/T
CONSOLES				
TELEPHONES:				
Single-line TNs				
Multi-line TNs				
TRUNKS:				
2-way				
1-way in				
1-way out				
IP Peer Virtual Trunks				
- SIP				
- H.323				
DID				
TIE				
CCSA				
InWATS				
OutWATS				
FX				
Private line				
Dial dictation				
Paging				
RAN				
AIOD (Automatic ID of Outward Dialing)				
DTI				
E&M 2W				
E&M 4W				

Line CCS/T __________

Total trunk CCS/T __________

Intra-CCS/T __________

553-AAA2159

Worksheet 11:Total load

LINE, TRUNK, AND CONSOLE USAGE

CUSTOMER ____________________ DATE ________

One sheet for each customer for cutover, 2-year, and 5-year interval.
One sheet for the system cutover, 2-year, and 5-year interval.

LINE USAGE:

Single-line TNs ______ X ______ CCS/T = ______ CCS

Multi-line TNs ______ X ______ CCS/T = ______ CCS

TOTAL LINE LOAD = ______ CCS

TRUNK USAGE:

Trunk route	Number of TNs accessing route		CCS/T per trunk route		Total CCS load per trunk route	
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS
______	______	X	______	=	______	CCS

TOTAL TRUNK LOAD = ______ CCS

CONSOLE USAGE:

Number of consoles ______ X 30 CCS = ______ TOTAL CONSOLE LOAD

DIGITONE RECEIVERS:

Table ______ Number of DTRs ______ TOTAL DTR LOAD ______ CCS

TOTAL LOAD ______ CCS

553-4042

Worksheet 12:Network loops

NETWORK LOOP CALCULATION

CUSTOMER ______________________________________ DATE ___________

One sheet for each customer. One sheet for the complete system.

	Total load		CCS per load		Number of loops	Round to next highest number
Cutover	________	÷	________	=	________	________
2-year	________	÷	________	=	________	________
5-year	________	÷	________	=	________	________

Number of network loops required at 2 years = ________

Number of network groups required at 2 years (use table below) = ________

Number of network groups	Maximum number of voice loops	Without Digitone trunks 744/560 CCS/loop	With Digitone trunks 720/540 CCS/loop
1	28	20 832 / 15 680	20 160 / 15 120
2	56	41 664 / 31 360	40 320 / 30 240
3	84	62 496 / 47 040	60 480 / 45 360
4	112	83 328 / 62 720	80 640 / 60 480
5	140	104 160 / 78 400	100 800 / 75 600
6	168	124 992 / 94 080	120 960 / 90 720
7	196	145 824 / 109 760	141 120 / 105 840
8	224	166 656 / 125 440	161 280 / 120 960

Note 1: The table above is based on an 85 percent utilization level.

For superloops, the *maximum* CCS/loop is 875 without Digitone trunks, 848 with Digitone trunks. Using the 85 percent utilization level, the CCS/loop is 744 without Digitone trunks, 720 with Digitone trunks.

For regular loops, the *maximum* CCS/loop is 660 without Digitone trunks, 560 with Digitone trunks. Using the 85 percent utilization level, the CCS/loop is 560 without Digitone trunks, 540 with Digitone trunks.

Note 2: At high traffic levels the CPU capacity needs to be calculated to determine whether there is sufficient capacity to process the given load.

553-AAA5361

Worksheet 13:Network loop balancing

BALANCING NETWORK LOOPS OVER NETWORK GROUPS

CUSTOMER____________________________________ DATE __________

One sheet for the complete system.

CUSTOMER	NETWORK GROUP 0	NETWORK GROUP 1	NETWORK GROUP 2	NETWORK GROUP 3	NETWORK GROUP 4	NETWORK GROUP 5	NETWORK GROUP 6	NETWORK GROUP 7

553-AAA0359

Worksheet 14:IPE card calculations

IPE CARD CALCULATIONS

CUSTOMER__________________________________ DATE __________

One for the complete system at cutover, 2-year, and 5-year interval.

NUMBER OF:	CUTOVER	2-YR	5-YR
Digital line cards = (Digital line ports + number of M2250 consoles) ÷ 16			
Analog line cards = Analog ports ÷ 16			
Analog message waiting line cards = Analog ports with message waiting ÷ 16			
Universal trunk cards = CO/DID/RAN/paging trunks ÷ 8			
2-W E&M/DX/paging trunks ÷ 2			
E&M trunk cards = E&M/paging/dictation trunks ÷ 4			
VGMC cards =			
Application cards =			
TOTAL CARDS			

IPE MODULE CALCULATIONS:

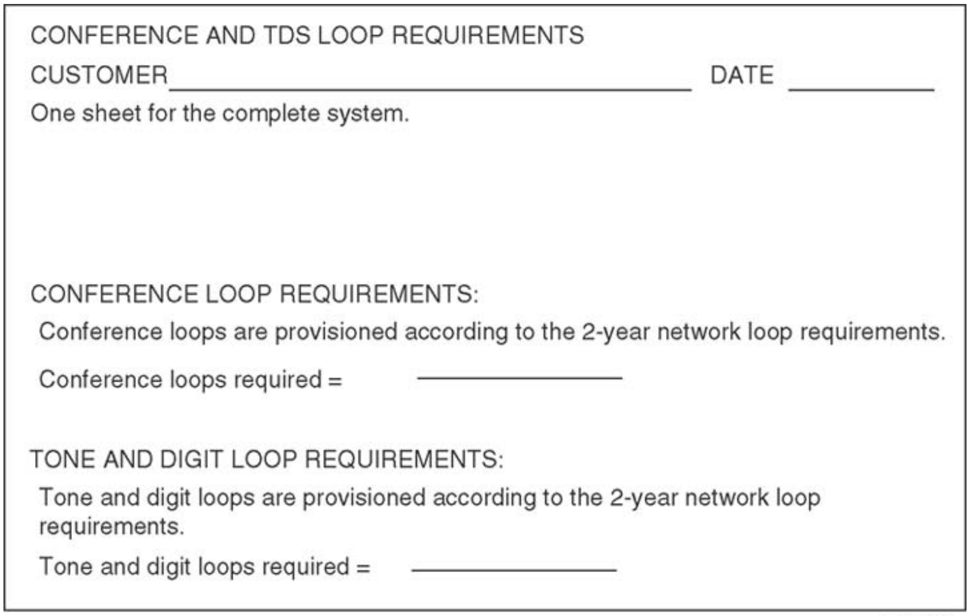
Use the total cards required at 2 years to determine the number of IPE Modules to be provisioned at cutover.

IPE Modules required = Total cards (round to next higher number) ÷ 8.5

NUMBER OF IPE MODULES REQUIRED AT CUTOVER ____________

553-AAA2118

Worksheet 15:Conference and TDS loop requirements


CONFERENCE AND TDS LOOP REQUIREMENTS

CUSTOMER________________________________ DATE __________

One sheet for the complete system.

CONFERENCE LOOP REQUIREMENTS:

Conference loops are provisioned according to the 2-year network loop requirements.

Conference loops required = __________

TONE AND DIGIT LOOP REQUIREMENTS:

Tone and digit loops are provisioned according to the 2-year network loop requirements.

Tone and digit loops required = __________

553-4046

Worksheet 16: Resource calculations

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)

Input parameters	Input configuration data
Hold time in seconds (AHT_{XX})for telephone to telephone, trunk to trunk, telephone to trunk, trunk to telephone	

Table 112: Worksheet 16a Media Card 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}
(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)

Item	Calculation formula
(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)	$= (R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$
(23) Total calls (T_{CALL})	$= 0.5 \times T_{CCS} \times 100 \div WAHT$ = 0.5 × (15) × 100 ÷ (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})	$= C_{TT} \times V^2 \times V_H \times V_S \times 2 \times 2$

Item	Calculation formula
(26) Originating / Outgoing calls (C_{ST})	= T_{CALL} × O (outgoing ratio)
• (a) UNIStim IP to VT calls (C_{STIV})	= C_{ST} × P_U × V
• (b) UNIStim IP to TDM calls (C_{STID})	= C_{ST} × P_U × (1 - V)
• (c) TDM telephone to VT calls (C_{STDV})	= C_{ST} × (1 - P_{IP}) × V
• (d) TDM telephone to TDM trunk calls (C_{STDD})	= C_{ST} × (1 - P_{IP}) × (1 - V)
• (e) SIP Line to VT calls (C_{STSV})	= C_{ST} × P_S × V
• (f) SIP Line to TDM trunk calls (C_{STSD})	= C_{ST} × P_S × (1 - V)
(27) Terminating / Incoming calls (C_{TS})	= C_{TS} × I (incoming ratio)
• (a) VT to TDM telephone calls (C_{TSVD})	= C_{TS} × V × (1 - P_{IP})
• (b) VT to UNIStim IP telephone calls (C_{TSVI})	= C_{TS} × V × P_U
• (c) TDM trunk to UNIStim IP telephone calls (C_{TSDI})	= C_{TS} × (1 - V) × P_U
• (d) TDM trunk to TDM telephone calls (C_{TSDD})	= C_{TS} × (1 - V) × (1 - P_{IP})
• (e) VT to SIP Line telephone calls (C_{TSVS})	= C_{TS} × V × P_S
• (f) TDM trunk to SIP Line telephone calls (C_{TSDS})	= C_{TS} × (1 - V) × P_S

Table 113: Worksheet 16b DSP and Media Card calculation

Item	Calculation formula
(1) Calls requiring DSP resources (C_{DSP})	= C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI} + C_{1SIP} + C_{STSD} + C_{TSDS} + ((2 × C_{NOIP}) × ELC_P)
(2) DSP CCS for general traffic (CCS_{DSP})	= C_{DSP} × WAHT ÷ 100
(3) DSP cards for general traffic	= CCS_{DSP} ÷ 794

Item	Calculation formula
(4) DSP channels for conference	= Total_telephones × P_{IP} × r_{CON} × 0.4
(5) DSP channels for applications	= (a) + (b) + (c) + (d) + (e) + (f) + (g) + (h)
• (a) CallPilot	= Number of CallPilot ports × P_{IP}
• (b) MIRAN	= Number of MIRAN ports × P_{IP}
• (c) MICB	= Number of MICB ports × P_{IP}
• (d) MIPCB	= Number of MIPCB ports × P_{IP}
• (e) MICA	= Number of MICA ports × P_{IP}
• (f) MIVS	= Number of MIVS ports × P_{IP}
• (g) BRI	= Number of BRI ports × P_{IP}
• (h) Agent greeting ports	= Number of Agent greeting ports × P_{IP}
(6) Total DSP channels	= [(3) × 32] + (4) + (5)
(7) Total Media Cards	= Roundup [(6) ÷ 32]

Table 114: Worksheet 16c 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} × V_S
(3) H.323 Virtual Trunk calls	C_{VT} × 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.	

Worksheet 17: Equipment summary

EQUIPMENT SUMMARY

CUSTOMER ______________________________ DATE __________

One sheet for the complete system.

	QUANTITY	BASED ON
Line and trunk cards		Cutover
DTR loops		2 year
Unprotected memory cards		2 year
Protected memory cards		2 year
Conference loops		2 year
TDS loops		2 year
CPUs		Cutover
IPE modules		2 year
Network loops (except conference and TDS)		2 year
Network groups		2 year
Voice Gateway Media Cards		2 year
Signaling Servers		2 year
Application cards		2 year

553-AAA2160

Worksheet 18: Network loop traffic capacity

	Column A			Column B (Loops)
TDS/CON Loops	One card (2 loops) per Network Module*			____________
BLOCKING:				
XNET Loop	Admin. Telephones	______ × 6 =	______CCS	
	Non-ACD trunks	+ ______ × 26=	______CCS	
	Subtotal	=	______÷ 875	=_______(N_{0x})

Column A			Column B (Loops)
NON_BLOCKING:			
XNET	Agent Telephones	___________	
	Supervisor Telephones	+ ___________	
	ACD Analog and RAN Trunks	+ ___________	
	Subtotal	= _______ ÷ 30	=________(N_1)
DTI Trunks		= _______ ÷ 24	=_______(N_{2d})
PRI Trunks		___________	
		+ 2	
		= _______ ÷ 24	=_______(N_{2p})
Music Ports		= _______ ÷ 30	=_______(N_{31})
Applications		_______ ÷ 24	=_______(N_{32})
Total loops (Sum of entries under column B)			=________(N_L)
Round up all calculations to the next integer. *Iterative procedure may be needed if the number of network modules required was not correctly estimated at the outset. Conclusion: N_L <= 32 use Avaya Communication Server 1000M SG/Meridian 1 Option 61C CP PIV 32 < N_L <256 use AvayaCommunication Server 1000M MG//Meridian 1 Option 81C CP PIV			

Worksheet 19: Physical capacity

The system type is determined by its required capacity. In general, for single group systems, the capacity limit is the number of loops (see Worksheet 18: Network loop traffic capacity on page 402), not the number of card slots.

Table 115: Worksheet 19a Card slot calculation

	Column A (Loop/card)		Column B (Slots)
TDS/CON	One/Network Module*		=__________
MUSic Loop	One TDS/CON provides one MUSic		=_____(N_{31})
NT8D04 XNET	Blocking Loops	_________(N_{0x})	
	nonblocking Loops	+_________(N_1)	
	Subtotal	=__________ ÷ 4	=_____(Sx)
NT5D12 DDP	2 DTI/PRI loops per slot		
NT5D97 DDP2	2 DTI2/PRI2 loops per slot		
NT6D80 MSDL	4 DCH ports or SDI ports per slot		
NT1P61 FNET	1 superloop per slot		
NT5D64 MCR	1 superloop per slot		
NTRB53 Clock Controller	1 slot per system		
I/O cards**	Must be $\geq S_x$		=__________
QPC720 DTI/PRI	= 2 × N_2, if no NT8D35 module; else = 0		=__________
Total # of card slots (sum of entries under column B)			=______(S_c)
Conclusion: $S_c \leq 16$ use Avaya Communication Server 1000M SG/Meridian 1 Option 61C CP PIV 16 < Sc use Avaya Communication Server 1000M MG/Meridian 1 Option 81C CP PIV			
Round up all calculations to the next integer. Set negative loop numbers to zero. *Iterative procedure may be needed, if the number of modules to use was not clear at the outset. **Refer to Table 4: Intelligent Peripheral Equipment on page 66 to find the number of I/O cards needed for applications.			

Worksheet 20: Signaling Server calculation

User input (modified if necessary from previous calculations)	
Number of UNIStim IP Phones in the system	__________
Number of SIP Line Phones in the system	__________

User input (modified if necessary from previous calculations)		
Number of Virtual Trunks - Number of SIP Virtual Trunks - Number of H.323 Virtual Trunks	__________ __________	 __________
Number of calls involving at least one UNIStim IP Phone (C_{UIP})	__________	
Number of calls involving at least one SIP Line Phone (C_{SIP})	__________	
Number of calls involving H.323 Virtual Trunks (HC_{VT})	__________	
Number of calls involving SIP Virtual Trunks (SC_{VT})	__________	
Endpoints served by this NRS (NRE)	__________	
Number of NRS entries (CDP + UDP + ...) (NRD)	__________	
Virtual Trunks from other endpoints served by this NRS (VT_{NET})	__________	
NRS alternate (NRA)	Yes/No	
TPS redundancy required (TPSA)	Yes/No	
H.323 Gateway alternate (GWA)	Yes/No	
SIP Gateway alternate (GSA)	Yes/No	
PD/CL/RL feature available to IP telephones	Yes/No	
PD/CL/RL feature sharing database with other traffic	Yes/No	

To continue Signaling Server calculation, see Signaling Server calculations on page 280.

Worksheet 21: Real time calculation

The variable values in many of the following tables are calculated in Worksheet 16 (Resource Calculations).

Table 116: Worksheet 21a: 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 = $C_{2sip} \div T_{CALL}$

Item	Calculation formula
(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	$P\_VTtoL = C_{TSVD} \div T_{CALL}$
(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	$PF = (P\_UIPtoUIP \times f_1) + (P\_UIPtoL \times f_2) + (P\_LtoL \times f_3) + (P\_VTtoTr \times f_4) + (P\_TrtoTr \times f_5) + (P\_VhtoVs \times f_6) + (P\_UIPtoVT \times f_7) + (P\_UIPtoTr \times f_8) + (P\_LtoVT \times f_9) + (P\_LtoTr \times f_{10}) + (P\_VTtoL \times f_{11}) + (P\_VTtoUIP \times f_{12}) + (P\_TrtoUIP \times f_{13}) + (P\_TrtoL \times f_{14}) + (P\_SIPtoSIP \times f_{15}) + (P\_SIPtoUIP \times f_{16}) +$

Item	Calculation formula
	(P_SIPtoL × f_{17}) + (P_SIPtoVT × f_{18}) + (P_SIPtoTr × f_{19}) + (P_VTtoSIP × f_{20}) + (P_TrtoSIP × f_{21})
(23) Error_term (basic features: forward/ transfer/conference/waiting)	Error_term = 0.25
(24) System EBC	SEBC = (T_{CALL} × (1 + PF + Error_term))

Table 117: Worksheet 21a: Feature and application EBC calculation

Item	Calculation formula
ACD calls	C_{ACD} = (L_{ACD} + L_{IPACD}) × 100 ÷ AHT_{AGT}
ACD	ACDEBC = C_{ACD} × (1 - %Symposium) × f_{ACD}
Symposium	SymposiumEBC = %Symposium × C_{ACD} × f_{SYM}
CallPilot	CallPilotEBC = (CP1 + CP2) × 100 ÷ AHT_{CP} × f_{CP}
Internal CDR	InternalCDR_EBC = C_{SS} × f_{ICDR}
Incoming CDR	IncomingCDR_EBC = C_{TS} × f_{CCDR}
Outgoing CDR	OutgoingCDR_EBC = C_{ST} × f_{OCDR}
Tandem CDR	TandemCDR_EBC = C_{TT} × f_{TAN}
Integrated Conference Bridge	MICB_EBC = number of Integrated Conference Bridge ports × $Appl_{CCS}$ × 100 ÷ AHT_{MICB} × f_{MICB}
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}) × f_{CPND}
ITG Trunk (only 11C,61C,81C)	ITG_EBC = [(ITG ports × 28) × 100 ÷ 150] × f_{ITG}

Item	Calculation formula
Converged Desktop	CD_EBC = (C_{SS} × 0.1 + C_{TT} + C_{ST} + C_{TS}) × r_{DTP} × f_{DTP}
Microsoft Converged Office	MO_EBC = (C_{SS} × 0.1 + C_{TT} + C_{ST} + C_{TS}) × mop × f_{MO}
IP Security	IPSEC_EBC = (C_{SS} + C_{TT} + C_{ST} + C_{TS}) × 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 × $f_{mobileX}$
Extend Local Calls (ELC)	ELC_EBC = C_{SS} × ELC_P x 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 + ITG_ELC + CD_EBC + MO_EBC + IPSEC_EBC + MC3100_EBC + MobileX_EBC + ELC_EBC

Table 118: Worksheet 21c: Real Time Usage calculation

Item	Calculation formula
Real Time Usage	RTU = (SEBC + FAEBC) ÷ Rated_EBC × 100 where Rated_EBC is the rated capacity of the CPU from Table 54: Real-time capacity (EBC) by system on page 218

Appendix B: Reference tables

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Trunk traffic Erlang B with P.01 Grade-of-Service

Table 119: 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	602	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 120: 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 121: 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 122: 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 123: 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

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
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 124: 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

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
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 125: 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

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
12	391	145	27	1337	495
13	448	166	28	1405	520
14	505	187	29	1472	545
15	562	208	30	1543	571
16	624	231			
See "Step 5: Calculate Digitone receiver requirements" for Model 4 assumptions.					

Digitone receiver load capacity 6 to 15 second holding time

Table 126: Digitone receiver load capacity 6 to 15 second holding time

	Average holding time in seconds									
Number of DTRs	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

	Average holding time in seconds									
Number of DTRs	6	7	8	9	10	11	12	13	14	15
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
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 127: Digitone receiver load capacity 16 to 25 second holding time

	Average holding time in seconds									
Number of DTRs	**16**	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**
1	0	0	0	0	0	0	0	0	0	0
2	2	2	2	2	2	2	2	2	2	2
3	8	8	8	8	8	8	8	8	8	8
4	18	18	18	18	18	17	17	17	17	17
5	31	31	31	30	30	30	30	30	30	29
6	47	46	46	45	45	45	45	44	44	44
7	64	63	63	62	62	62	61	61	61	60
8	83	82	82	81	80	80	79	79	79	78
9	103	102	101	100	100	99	99	98	98	97
10	125	123	122	121	121	120	119	119	118	118
11	147	145	144	143	142	141	140	140	139	138
12	170	168	167	166	165	164	163	162	161	160
13	193	192	190	189	188	186	185	184	184	183
14	218	216	214	213	211	210	209	208	207	206
15	243	241	239	237	236	234	233	232	231	230
16	268	266	264	262	260	259	257	256	255	254
17	294	292	290	288	286	284	283	281	280	279
18	322	319	317	314	312	311	309	308	306	305
19	347	344	342	339	337	335	334	332	331	329
20	374	371	368	366	364	361	360	358	356	355
21	402	399	396	393	391	388	386	385	383	381
22	431	427	424	421	419	416	414	412	410	409
23	458	454	451	448	445	442	440	438	436	434
24	486	482	478	475	472	470	467	465	463	461

	Average holding time in seconds									
Number of DTRs	16	17	18	19	20	21	22	23	24	25
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
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.										

Digitone receiver requirements Poisson 0.1 percent blocking

Table 128: 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

Number of DTRs	DTR load (CCS)	Number of DTRs	DTR load (CCS)
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
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 129: 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

Network loops required at 2 years	TDS loops required	Conference loops required
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 130: 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

DTR CCS	DTR ports	DTR CCS	DTR ports
372–398	20	1051–1084	42
399–427	21	1085–1118	43
428–456	22	1119–1153	44
457–487	23	1154–1188	45
1189–1223	46	1961–1995	68
1224–1258	47	1996–2030	69
1259–1293	48	2031–2065	70
1294–1329	49	2066–2100	71
1330–1365	50	2101–2135	72
1366–1400	51	2136–2170	73
1401–1435	52	2171–2205	74
1436–1470	53	2206–2240	75
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

DTR CCS	DTR ports	DTR CCS	DTR ports
2906–2940	95	3116–3465	101
Provisioning assumes an 11-second holding time.			

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