NORTEL

Nortel Communication Server 1000

Communication Server 1000M and Meridian 1 Small System Planning and Engineering

NN43011-220

Document status: Standard
Document version: 01.01
Document date: 30 May 2007

Revision history

May 2007

Standard 01.01. This document is issued to support Communication Server 1000 Release 5.0 This document contains information previously contained in the following legacy document, now retired: *Communication Server 1000M and Meridian 1: Small System Planning & Engineering (553-3011-120)*. No new content has been added for Communication Server 1000 Release 5.0. All references to Communication Server 1000 Release 4.5 are applicable to Communication Server 1000 Release 5.0.

Descriptions and procedures pertaining to IP applications in this document are provided for continuity for customers remaining on Communication Server Release 4.5. A stand-alone IP Trunk (ITG Trunk) configuration is the only IP application supported on the Meridian 1 Option 11C platform in Communication Server 1000 Release 5.0. For information on software-only upgrades, refer to *Meridian 1 Small System Software-only Upgrade (NN43011-459)*.

Systems described within this document that are configured with IP Phones or Signaling Servers using Communication Server 1000 Release 4.5 and want to upgrade to Communication Server 1000 Release 5.0 must be upgraded to Communication Server 1000E with a Common Processor Pentium Mobile (CP PM) call processor. For migrations to Communication Server Release 5.0, refer to:

- *Communication Server 1000E Upgrade - Option 11C Cabinet to CS 1000E (NN43041-464)*

- *Communication Server 1000E Upgrade - Option 11C Chassis to CS 1000E (NN43041-465)*

- *Communication Server 1000E Upgrade - CS 1000M Cabinet to CS 1000E (NN43041-466)*

- *Communication Server 1000E Upgrade - CS 1000M Chassis to CS 1000E (NN43041-467)*

July 2006

Standard 6.00 This document is up-issued to revise Phantom and Virtual loop information.

April 2006
Standard 5.00. This document is up–issued to include updated information.

January 2006
Standard 4.00. This document is up–issued to include SIP CTI / TR 87 information from the new Nortel Converged Office Implementation Guide (553-3001-025).

August 2005
Standard 3.00. This document is up–issued to support Communication Server 1000 Release 4.5. This document now contains a chapter on data network planning for VoIP.

September 2004
Standard 2.00. This document is up-issued to support Communication Server 1000 Release 4.0.

October 2003
Standard 1.00. This document is a new NTP for Succession 3.0. It was created to support a restructuring of the Documentation Library, which resulted in the merging of multiple legacy NTPs. This new document consolidates information previously contained in the following legacy documents, now retired:

- *Option 11C and 11C Mini Technical Reference Guide* (553-3011-100) (Content from *Option 11C and 11C Mini Technical Reference Guide* (553-3011-100) also appears in *Communication Server 1000M and Meridian 1 Small System Overview (NN43011-110)*, *Transmission Parameters Reference (NN43001-282)*, and *Circuit Card Reference (NN43001-311)*.)

- *Option 11 Planning and Installation Guide* (553-3021-210) (Content *from Option 11 Planning and Installation Guide* (553-3021-210) also appears in *Communication Server 1000M and Meridian 1 Small System Overview (NN43011-110)* and *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.)

- *Option 11C Mini Planning and Installation Guide* (553-3021-209) Content from *Option 11C Mini Planning and Installation Guide* (553-3021-209) also appears in *Communication Server 1000M and Meridian 1 Small System Overview (NN43011-110)* and *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.)

Contents

Nortel Communication Server 1000
Communication Server 1000M and Meridian 1 Small System Planning and Engineering
NN43011-220   01.01   Standard
Release 5.0   30 May 2007

Introduction

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*   *Communication Server 1000E Upgrade - CS 1000M Cabinet to CS 1000E (NN43041-466)*

*   *Communication Server 1000E Upgrade - CS 1000M Chassis to CS 1000E (NN43041-467)*

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

Subject

This Nortel Technical Publication (NTP) is a reference tool for activities surrounding the planning and engineering of a Small System.

> ⚠️ **WARNING**
>
> Before a Small System can be installed, a network assessment **must** be performed and the network must be VoIP-ready.
>
> If the minimum VoIP network requirements are not met, the system will not operate properly.
>
> For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Note on legacy products and releases

This NTP contains information about systems, components, and features that are compatible with Nortel Communication Server 1000 Release 4.5 software. For more information on legacy products and releases, click the **Technical Documentation** link under **Support & Training** on the Nortel home page:

www.nortel.com

Applicable systems

This document applies to the following systems:

- Communication Server 1000M Chassis (CS 1000M Chassis)
- Communication Server 1000M Cabinet (CS 1000M Cabinet)
- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet

System migration

When particular Meridian 1 systems are upgraded to run CS 1000 Release 5.0 and configured to include a Signaling Server, they become CS 1000M systems. Table 1 "Meridian 1 systems to CS 1000M systems" (page 12) lists each Meridian 1 system that supports an upgrade path to a CS 1000M system.

Table 1
Meridian 1 systems to CS 1000M systems

This Meridian 1 system...	Maps to this CS 1000M system
Meridian 1 PBX 11C Chassis	CS 1000M Chassis
Meridian 1 PBX 11C Cabinet	CS 1000M Cabinet

Note the following:

- When an Option 11C Mini system is upgraded to run CS 1000 Release 5.0 software, that system becomes a Meridian 1 PBX 11C Chassis.

- When an Option 11C system is upgraded to run CS 1000 Release 5.0 software, that system becomes a Meridian 1 PBX 11C Cabinet.

For more information, see *Meridian 1 Small System Software-only Upgrade (NN43011-459)*.

Intended audience

This document is intended for individuals responsible for planning and engineering the Small System. The engineer may be an employee of the end-user customer, a third-party consultant, or a distributor.

Other persons who may be interested in this information, or find it useful, are Sales and Marketing, Service Managers, Account Managers, Field Support, Product Management, and Development.

Conventions

Terminology

In this document, the following systems are referred to generically as "system":

- Communication Server 1000M (CS 1000M)
- Meridian 1

The following systems are referred to generically as "Small System":

- Communication Server 1000M Chassis (CS 1000M Chassis)
- Communication Server 1000M Cabinet (CS 1000M Cabinet)
- Meridian 1 PBX 11C Chassis
- Meridian 1 PBX 11C Cabinet

The following systems are referred to generically as "Chassis system":

- Communication Server 1000M Chassis (CS 1000M Chassis)
- Meridian 1 PBX 11C Chassis

The following systems are referred to generically as "Cabinet system":

- Communication Server 1000M Cabinet (CS 1000M Cabinet)
- Meridian 1 PBX 11C Cabinet

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Spares Planning (NN43001-253)*
- *Converging the Data Network with VoIP Fundamentals (NN43001-260)*
- *Transmission Parameters Reference (NN43001-282)*
- *Circuit Card Reference (NN43001-311)*
- *Features and Services Fundamentals (NN43001-106)*
- *Software Input Output Administration (NN43001-611)*
- *IP Line Fundamentals (NN43001-500)*
- *Telephones and Consoles Fundamentals (NN43001-567)*
- *Software Input Output Reference - Maintenance (NN43001-711)*
- *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*
- *Meridian 1 Small System Software-only Upgrade (NN43011-459)*
- *Communication Server 1000M and Meridian 1 Small System Maintenance (NN43011-700)*

Online

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

www.nortel.com

CD-ROM

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

Overview of the engineering process

Contents

This section contains information on the following topics:

Introduction

> **WARNING**
>
> Before a Small System can be installed, a network assessment must be performed and the network **must** be VoIP-ready.
>
> If the minimum VoIP network requirements are not met, the system will not operate properly.
>
> For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

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

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

Engineering a new system

Figure 1 "Engineering a new system" (page 17) 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. Figure 2 "Engineering a new system" (page 18) further illustrates the engineering process.

Engineering a system upgrade

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

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 with the capacity chart to determine whether the corresponding capacity has been exceeded.

Figure 1
Engineering a new system


```
                        From Customer

                              |
                              v

                   Perform Network
                   Assessment to Determine
                   VoIP Readiness                          Agent

                              |   Customer
                              v   Requirements

                        Propose                            Sales,
                        System            <--------        Marketing,
                        Solution                           Distributor

                              |   Functional
                              v   Requirements

                   Match Features to                       Customer
                   Functions, Propose     <--------        Service
                   System Configuration                    Representative

                              |   System
                              v   Proposal

        Engineer                        NNEC                System
        the                             (Enterprise        Engineer
        System                          Configurator)

                              |   Engineered
                              v   System

                        Assign                             Customer
                        Equipment                          Service
                                                           Representative

                              |
                              v

                   Perform Network
                   Assessment if Network
                   Changed Since
                   Last Assessment

                              |   Bay Face
                              v   Layout

                        Install                            Technician
                        the
                        System

                              |   Installed
                              v   System

                        To Customer                        553-AAA1539
```

Figure 2
Engineering a new system

System
Proposal

Choose a(nother)
Capacity Worksheet
from "Worksheets"

Go to first (next)
line on
Worksheet

Is
Input Value
Known? — No → Follow Instructions "System
capacities" or Detailed
Guidelines in "Application
engineering"

Yes

Read and
Execute
Procedure

Enter
Input Value
in Worksheet

Is this
the last line in
worksheet? — No

Yes

Compute
Capacity Impact
Using Worksheet

Note: Refer to "System
capacities" for an
explanation of the Capacity
Algorithm

Compare Impact
with
Capacity Chart

Capacity
Exceeded? — No → All
Worksheets
Completed? — No

Yes

Yes

Feedback to CSR
or Sales

Engineered
System

553-0041

Nortel Enterprise Configurator

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

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

NNEC runs on the user's Windows®-based or MacOS® personal computer. It uses standard browser and Microsoft Office® applications. For details on computer system requirements and for user instructions, refer to the Nortel web site.

NNEC 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 meets the customer's capacity requirements.

Where applicable, this document references the NNEC inputs that correspond to parameters being described.

Data network planning for VoIP

Contents

This section contains information on the following topics:

Introduction

> **WARNING**
> Before a Small System can be installed, a network assessment **must** be performed and the network must be VoIP-ready.
>
> If the minimum VoIP network requirements are not met, the system will not operate properly.
>
> For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

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 Small System.

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

Data network planning for VoIP

Consider the following when planning the network:

- system network requirements (for ELAN and TLAN subnets)

- basic data network requirements for Call Server to Media Gateway connections

  — 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 Small System on a data network

To deploy the Small System on a data network, consider the following details and refer to *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.

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

Core network planning

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

Step	Action
1	Call Server to Media Gateway network
2	ELAN (Management LAN) subnet
3	TLAN (Voice LAN) subnet

> *Note:* 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.

—End—

100BaseTx IP connectivity

Between the Call Server and the Media Gateway, the CS 1000 systems support 100BaseTx IP point-to-point connectivity or campus data network connectivity. Campus data network connectivity is provided through IP daughterboards in the Call Server and the Media Gateway.

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

Campus network system requirements

The following campus network requirements are necessary:

- The ELAN subnet and the TLAN subnet must be separate.

- ELAN subnet applications must be on the same subnet. This includes the Voice Gateway Media Cards, which must be on the same ELAN subnet.

- Voice Gateway Media Cards in the same node must be on the same TLAN subnet.

- Use of the VLAN concept is a practical way to maintain the same subnet for remote locations.

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

- 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.

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Regulatory information

Contents

This section contains information on the following topics:

System approval

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

> *Note:* 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 *Communication Server 1000*. Similarly, previous naming conventions utilizing the terms *Meridian* and *Option* have been harmonized to use the term *Meridian 1 PBX*. Product names based on earlier naming conventions 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 Communication Server 1000M, Communication Server 1000S, and Communication Server 1000E systems, as well as the Meridian, Succession 1000, and CSE 1000 systems.

Electromagnetic compatibility

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

Note: 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 2 "EMC specifications for Class A devices" (page 28) lists the Electromagnetic Compatibility (EMC) specifications for the system.

Table 2
EMC specifications for Class A devices

Jurisdiction	Standard	Description
United States	FCC CFR 47 Part 15	FCC Rules for Radio Frequency Devices (see Note 1)
Canada	ICES-003	Interference-Causing Equipment Standard: Digital Apparatus

*Note 1 a:*FCC CFR 47 Part 15.21 statement:
"Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, can cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user is required to correct the interference at his own expense."

*Note 1b:*The user should not make changes or modifications not expressly approved by Nortel. Any such changes could void the user's authority to operate the equipment.

*Note 2:*EN 55022/CISPR 22 statement:
"WARNING
This is a class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures."

Jurisdiction	Standard	Description
Europe	EN 55022/CISPR 22	Information technology equipment — Radio disturbance characteristics — Limits and methods of measurement (see Note 2)
	EN 55024	Information technology equipment — Immunity characteristics — Limits and methods of measurement
	EN 61000-3-2	Limits for harmonic current emissions (equipment input current <= 16 A per phase)
	EN 61000-3-3	Limitation of voltage fluctuations and flicker in low-voltage supply systems for equipment with rated current <= 16 A
Australia	CISPR 22/AS/NZS 3548	Limits and methods of measurement of radio disturbance characteristics of information technology equipment (see Note 2)
Korea	KN22	Information technology equipment — Radio disturbance characteristics — Limits and methods of measurement
	KN24	Information technology equipment — Immunity characteristics — Limits and methods of measurement
Taiwan	CNS 13438	Limits and methods of measurement of radio disturbance characteristics of information technology equipment

*Note 1 a:*FCC CFR 47 Part 15.21 statement:
"Note: This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, can cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user is required to correct the interference at his own expense."

*Note 1b:*The user should not make changes or modifications not expressly approved by Nortel. Any such changes could void the user's authority to operate the equipment.

*Note 2:*EN 55022/CISPR 22 statement:
"WARNING
This is a class A product. In a domestic environment this product 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.Each system cabinet or chassis has a label containing the FCC registration number and Ringer Equivalence Number (REN). For the Cabinet system equipment, the label is on the lower

left corner of each system cabinet. For the Chassis system equipment, the label is on the back of each system chassis. 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.

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

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

Radio and TV interference

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

Canadian and US network connections

Table 3 "Network connection specifications" (page 33) contains information that must be given to the local telephone company when ordering standard network interface jacks for the system.

Table 3 "Network connection specifications" (page 33) includes columns for system port identification, Facility Interface Code (FIC), Service Order Code (SOC), Uniform Service Order Code (USOC) jack identification, and associated Nortel equipment part numbers.

Table 3
Network connection specifications

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
MTS/WATS					
2-Wire, LSA, L-S (2-Wire, Local Switched Access, Loop-Start)	02LS2	9.0F	2.7A	RJ21X CA21X*	NT8D14
2-Wire, LSA, G-S (2-Wire, Local Switched Access, Ground-Start)	02GS2	9.0F	2.7A	RJ21X CA21X*	NT8D14
2-Wire, LSA, R-B (2-Wire, Local Switched Access, Reverse-Battery)	02RV2-T	9.0F	0.0B	RJ21X CA21X*	NT8D14
* RJ with CA for Canada					

Ports	Facility Interface Code	Service Order Code	REN	Network jacks	Manufacturer network interface port designation
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.

DenAn regulatory notice for Japan

取扱説明書

安全上のご注意

本取扱説明書「安全上のご注意」は以下のノーテル製品の取扱説明書の別紙であり、取扱説明書本文と不可分のものです。

- Communication Server 1000M Cabinet/Chassis
- Communication Server 1000S
- Communication Server 1000E
- Meridian 1 Option 11C
- Meridian 1 Option 11C Mini
- Media Gateway 1000
- Multimedia Communication Server 5100
- CallPilot 703t server
- Hospitality Messaging Server 400
- Media Processing Server 500
- Media Processing Server 1000

⚠ 警告

本製品を安全にご使用頂くため、以下のことにご注意ください。

● 接続ケーブル、電源コード、ACアダプタなどの部品は、必ず製品に同梱されております添付品または指定品をご使用ください。添付品・指定品以外の部品をご使用になると故障や動作不良、火災の原因となることがあります。

● 同梱されております付属の電源コードを他の機器には使用しないでください。上記注意事項を守らないと、死亡や大怪我など人身事故の原因となることがあります。

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System and site requirements

Contents

This section contains information on the following topics:

> **WARNING**
> Before a Small System can be installed, a network assessment **must** be performed and the network must be VoIP-ready.
>
> If the minimum VoIP network requirements are not met, the system will not operate properly.
>
> For information on the minimum VoIP network requirements and converging a data network with VoIP, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Before you install a Small System, make sure that the site meets all environmental, grounding, power, and cross-connect terminal requirements.

Environmental requirements

The environment of a Small System operates must meet the following operating conditions:

- The room must be clean and well ventilated. Each chassis can dissipate up to 370 watts of power; each cabinet can dissipate up to 500 watts of power in the form of heat (1700 BTU [1800 kJ] per hour). There must be enough ventilation in the equipment room to maintain the temperature at an acceptable level.

- For installed chassis, maintain the temperature between 0° and 45° C (32° and 113° F). For cabinets, the temperature is maintained between:

  — 0° and 45° C (32° and 113° F) when the cabinets are mounted side-by-side.

  — 0° and 35° C (32° and 95° F) when the cabinets are mounted one above the other.

- Maintain the humidity between 5% and 95% non-condensing.

- Select a location for installing the equipment that is not subject to constant vibration.

- Locate the equipment at least 12 ft (3660 mm) away from sources of electrostatic, electromagnetic, or radio frequency interference. These sources can include:

  — power tools

  — appliances (such as vacuum cleaners)

  — office business machines (such as copying machines)

  — all electric motors

  — electrical transformers

Earthquake bracing requirements

> **ATTENTION**
>
> **IMPORTANT!**
> The following earthquake bracing guidelines meet the requirements for the state of California specifications in the United States. Other areas or countries can have different requirements.

Note: The earthquake bracing method for the Small System does not guarantee that the system will continue to operate during or after an earthquake.

> **WARNING**
> For earthquake bracing, you must install the Small System on a wall in a vertical position.

To earthquake-brace your system, use a piece of 3/4-in. (20-mm) plywood as a backboard. Fasten the plywood to the wall with a minimum of six fasteners. Refer to Table 4 "Minimum fastener requirements" (page 39) for a description of the appropriate fasteners. Fasten the chassis to the backboard.

Table 4
Minimum fastener requirements

Type of wall	Fasteners	
Wooden studs	#10 wood screws	Minimum 1 in. embedment in wood studs
Metal studs	# 14 sheet metal screws	Minimum 1 in. embedment in metal studs
Concrete (2000 PSI)	1/4 in. HILTI KB-II	Minimum 1 1/8 in. embedment
Masonry	1/4 in. Ramset Redhead Dynabolt sleeve anchor	

Table 5 "Minimum wall requirements for stud construction" (page 39) identifies the maximum acceptable wall height for different types of stud wall construction in areas prone to earthquakes.

Table 5
Minimum wall requirements for stud construction

Wall Studs	Spacing off center	Maximum Height of Wall
2 in. x 4 in. wooden studs	16 in. or 24 in.	10 ft
2 in. x 6 in. wooden studs	16 in. or 24 in.	16 ft
3 5/8 in. 20 gauge metal studs	16 in. or 24 in.	12 ft
3 5/8 in. 18 gauge metal studs	16 in. or 24 in.	16 ft

Fasten the mounting bracket for each chassis and cabinet to the piece of plywood with the five, 1-in. #14 screws supplied with the bracket.

Refer to the chapter on earthquake bracing in *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for a detailed procedure on earthquake bracing.

Grounding requirements

System grounding must be in accordance with ANSI/TIA/EIA-607 (Commercial Building and Bonding Requirements for Telecommunications Equipment) where discrepancies in this document exist.

⚠️	**WARNING** Failure to follow grounding recommendations can result in a system installation that is: • unsafe for personnel working on, or using, the equipment • not protected correctly from lightning or power transients • subject to service interruptions

Before you install a Small System and before you apply AC power, measure the impedance of the building ground reference. An ECOS 1023 POW-R-MATE, or similar meter, is acceptable for this purpose. If the ground path connected to the Small System has an impedance of 5 ohms or more, make better grounding arrangements. Make any improvements to the grounding system before you install the Small System.

⚠️	**DANGER** **DANGER OF ELECTRIC SHOCK** Never connect the Single Point Ground (SPG) conductor from the Small System to structural steel members or electrical conduit. Never tie this conductor to a ground source or grounded electrode that is not hard-wired to the building reference conductor.

The following are additional grounding requirements:

- Grounding requirements for the Small System are as follows:

  — The impedance of the link between the ground post of the system cabinets or chassis and the SPG to which it is connected must be less than 0.25 ohms.

  — Never connect the SPG conductor from the Small System to structural steel members or electrical conduit. In particular, never tie this conductor to a ground source or grounded electrode that is not hard-wired to the building reference conductor.

- Ground conductors for Small Systems:

  — must not be smaller than #6 AWG (#40 metric) at any point (see Table 6 "Area-specific grounding wire requirements" (page 41). This table provides a list of grounding wire requirements specific to some areas).

— must be routed through the same conduit as the phase conductors that serve the system.

— must not be smaller than any phase conductor in the same conduit.

— do not carry current under normal operating conditions.

- All ground conductors in the building must be hard-wired to the main ground reference.

- Avoid spliced conductors. Continuous conductors have lower impedance, and they are more reliable than spliced conductors.

- All conductors must terminate in a permanent way. Make sure all terminations are easily visible and available for maintenance purposes.

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

Table 6
Area-specific grounding wire requirements

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

DANGER
DANGER OF ELECTRIC SHOCK
For an installed Small System, the impedance of the link between the ground post of the chassis or cabinet and the SPG to which it connects must be less than 0.25 ohms.

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

Ground bus isolation (Canada and the United States)
According to the exception to article 384-20 in the United States National Electrical Code (NEC), a panel's ground bus can be isolated from the housing. This exception applies provided that the panel is not at the main

service entrance. This exception applies to some Canadian locations also. For more information about ground bus isolation, refer to local electrical codes.

> ⚠️ **DANGER**
> **DANGER OF ELECTRIC SHOCK**
> Do not isolate the ground bus from the housing unless permitted by local electrical codes. Do not perform work inside electrical panels unless you are a qualified electrician. Do not try to remove bonding conductors without approval from qualified personnel.

> ⚠️ **DANGER**
> **DANGER OF ELECTRIC SHOCK**
> Route ground conductors, between supply panels, through the same conduit as the supply conductors. This safety requirement is part of both the National Electrical Code (NEC) and the Canadian Electrical Code (CEC).

Single Point Grounding

Correct grounding of communications systems is necessary for protecting equipment from the hazards of surge and noise interference. The Single Point Grounding (SPG) method of protecting communications equipment is the Nortel standard for Small Systems. Refer to Figure 3 "Single Point Grounding" (page 44) for an illustration of Single Point Grounding.

The requirements for Single Point Grounding are in the following major categories: Safety, Protection, Electromagnetic Compatibility (EMC), Installation and Maintenance, Powering, and Advances in Technology.

Safety For a safe working environment, your grounding system must be able to dissipate unwanted surge energies, such as lightning on the outside plant. The grounding system must be designed so that fuses or breakers operate to disrupt the excessive current flow caused by a power fault.

Protection Correct grounding is a necessary component of the protection system for equipment. This grounding includes grounding for outside plant cable shields and protectors, and the grounds for framework, battery, and logic references.

EMC To make sure that there is good emission and susceptibility performance of the equipment, you must consider the EMC grounding requirements.

Installation and Maintenance   A grounding system is cost-effective to install and maintain when it is part of the initial electrical installation for the customer's premises. If you try to correct violations of national codes after the initial installation, it is both difficult and costly.

Powering   The grounding system must consider the power options for the equipment. The grounding system must consider if the equipment is backed up with an Uninterruptible Power Supply (UPS). Consider the grounding and powering of all equipment that is part of the telecommunications system as one large system. Perform the installation taking this information into consideration.

Advances in Technology   The component density on circuit cards continues to increase because of the miniaturization and multi-layering of printed circuit boards. The operating speeds of the integrated circuits are ever-increasing. Grounding provides protection for these components and is very important.

SPG configurations

The SPG of a system is the point at which telecommunications equipment bonds to the ground. A copper bus bar normally acts as the system SPG.

You can use any of the following bus bars as a system SPG:

- building principal ground, normally in a building with one floor
- floor ground bar, normally in buildings with more than one floor
- dedicated SPG bar bonded to the building grounding system
- a section of the battery return bar of the power plant

You can configure subsystems of a telecommunications system, such as groups of frames or equipment, as separate SPG entities connected in a star configuration to the system SPG. Figure 3 "Single Point Grounding" (page 44) provides a high-level schematic of this type of configuration.

Figure 3
Single Point Grounding
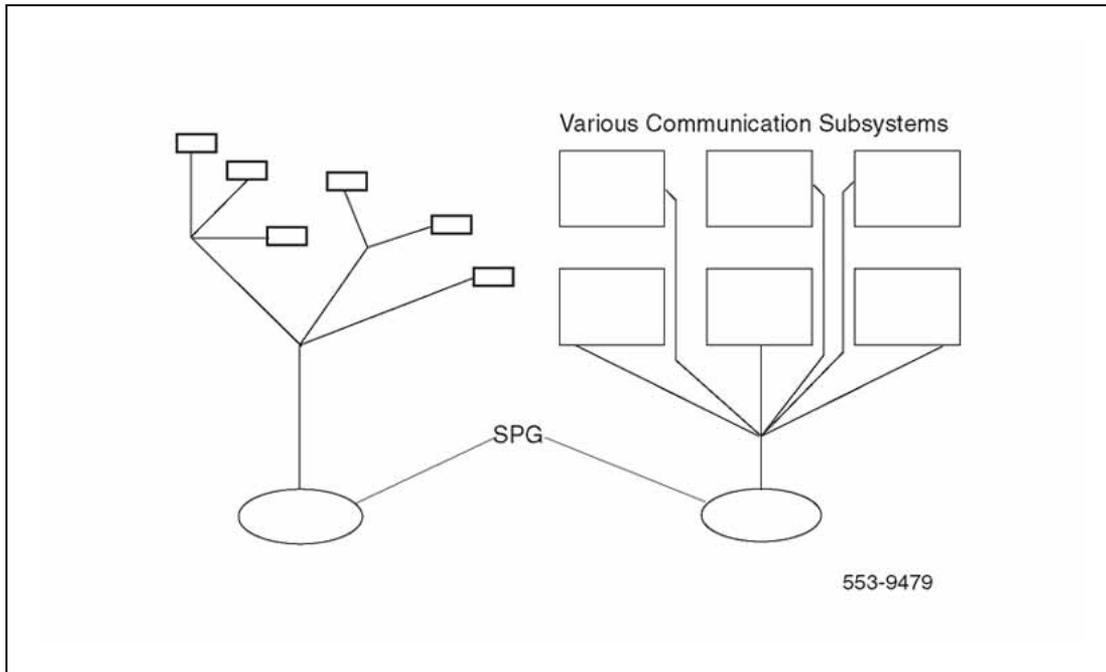
Figure 4 "ANSI/TIA/EIA-607 grounding schematic" (page 45) provides a more detailed schematic of telecommunications subsystems configured as separate SPG entities and connected to the building's SPG using the Telecommunications Main Grounding Bus bar (TMGB) and Telecommunications Grounding Bus bars (TGB).

Figure 4
ANSI/TIA/EIA-607 grounding schematic

553-AAA1160
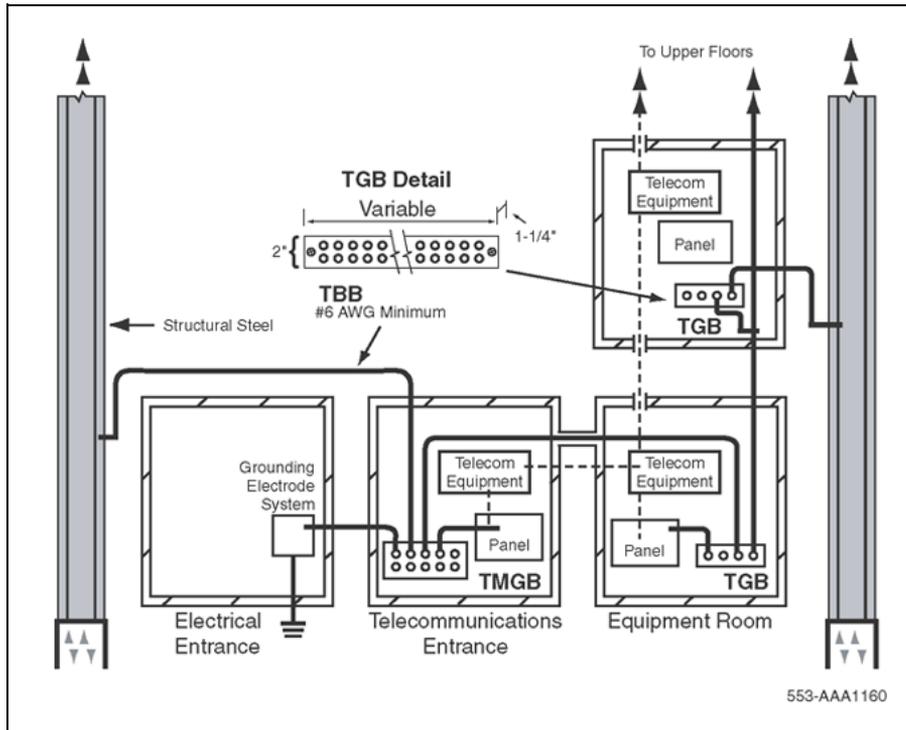
Grounding method

> **⚠ WARNING**
>
> To prevent ground loops, power all chassis and cabinets from the same dedicated power panel. Ground all chassis and cabinets to the power panel through the grounding block. Ground the chassis expander to the chassis.

The method of grounding used for Small Systems depends on whether the same service panel powers all chassis or cabinets.

The following grounding scenarios are possible:

1. Chassis system with one chassis

2. Cabinet system with one cabinet

3. Chassis system with more than one chassis, powered by the same service panel

4. Cabinet system with more than one cabinet, powered by the same service panel

5. Chassis system with more than one chassis, powered by different service panels

6. Cabinet system with more than one cabinet, powered by different service panels

A system with one chassis or cabinet, or multiple chassis or cabinets powered by one service panel

For each system chassis or cabinet, connect a #6 AWG (#40 Metric Wire Gauge) ground wire from the chassis or cabinet to the NTBK80 grounding block. See Table 6 "Area-specific grounding wire requirements" (page 41) for grounding wire requirements specific to some areas. Connect the grounding block to a ground source (the ground bus in the AC power service panel).

For a Chassis system, consider the chassis and the chassis expander as the same ground. Jumper the ground wire from the chassis expander to the chassis and then back to the grounding block.

Chassis or cabinets powered by different service panels

For each chassis or cabinet, connect a #6 AWG (#40 Metric Wire Gauge) ground wire from the chassis or cabinet to the NTBK80 grounding block. See Table 6 "Area-specific grounding wire requirements" (page 41) for grounding wire requirements specific to some areas. If any chassis or cabinet cannot be powered from the same service panel, ground it separately from the other chassis or cabinet back to the service panel that supplies it. Power each cabinet or chassis and chassis expander pair from the same service panel.

Note 1: If a chassis requires a separate ground, ground it using the same method that you use for a system with one chassis. A separately grounded cabinet is grounded the same as a single-cabinet system.

Note 2: In the UK, you can connect the grounding wire from the cabinet or chassis to an NTBK80 grounding block or through a Krone Test Jack Frame.

Grounding multiple pieces of equipment in a rack/equipment cabinet

You must ground each piece of equipment in a rack/equipment cabinet. If a piece of equipment does not have a ground lug, then ground the whole rack/equipment cabinet.

Conduit requirements

Conductive conduit linking panels and equipment are legal for use as a grounding network in most countries. For all system ground paths, use the correct size of insulated copper conductors routed inside conduit

when possible. A ground link that depends on conduit can defeat the improvements made by installing dedicated panels and transformers, for the following reasons:

- Personnel who service different equipment can separate conduit links. If this separation occurs between the Small System and the building ground reference, the conduit cannot provide a ground path. This situation is hazardous.

- Metal conduit often corrodes, especially at threaded connections. Corrosion increases resistance. This problem becomes worse when multiple links are involved. If you apply paint over the conduit, it is possible that the corrosion process will occur more quickly.

- Always fasten conduit to secure surfaces. Often, conduit is bolted to structural steel members, which can function as ground conductors to noisy equipment (for example, compressors and motors). The coupling of these noisy signals into the Small System grounding system can damage its performance. The resulting intermittent malfunctions can be difficult to trace.

Commercial power requirements

The Chassis system is available with AC power only. The Cabinet system is available in both AC-powered and DC-powered versions.

The optimal installation of the AC-powered Small System includes a direct connection to the electrical system in the building, provided some requirements are met. Refer to "AC-powered installation" (page 47) for detailed information. The chassis and chassis expander can share the same electrical breaker.

You can use an approved Isolation Transformer for AC-powered systems when meeting the optimum requirements is not possible or is too expensive. See "Alternative AC-powered installation" (page 51).

Refer to "Power consumption" (page 56) to determine the power consumption of the Chassis system.

With the DC-powered version of the Cabinet system, each cabinet is powered solely from a DC power source. See "DC-powered version (Cabinet system only)" (page 55) for detailed information.

AC-powered installation

Use a dedicated AC service panel with a Small System. Do not connect equipment that is not related to the Small System to this panel. Keep all lighting, fans, motors, air conditioning equipment, and such as "electrically separate" from the Small System as possible. The chassis and chassis expander can share the same AC breaker.

If other data communications equipment is in the same rack/equipment cabinet as the Small System, power each piece of equipment from a grounded outlet. The same service panel must service all outlets.

Figure 5 "Typical wiring plan" (page 48) shows a typical wiring plan.

Figure 5
Typical wiring plan

553-AAA2388
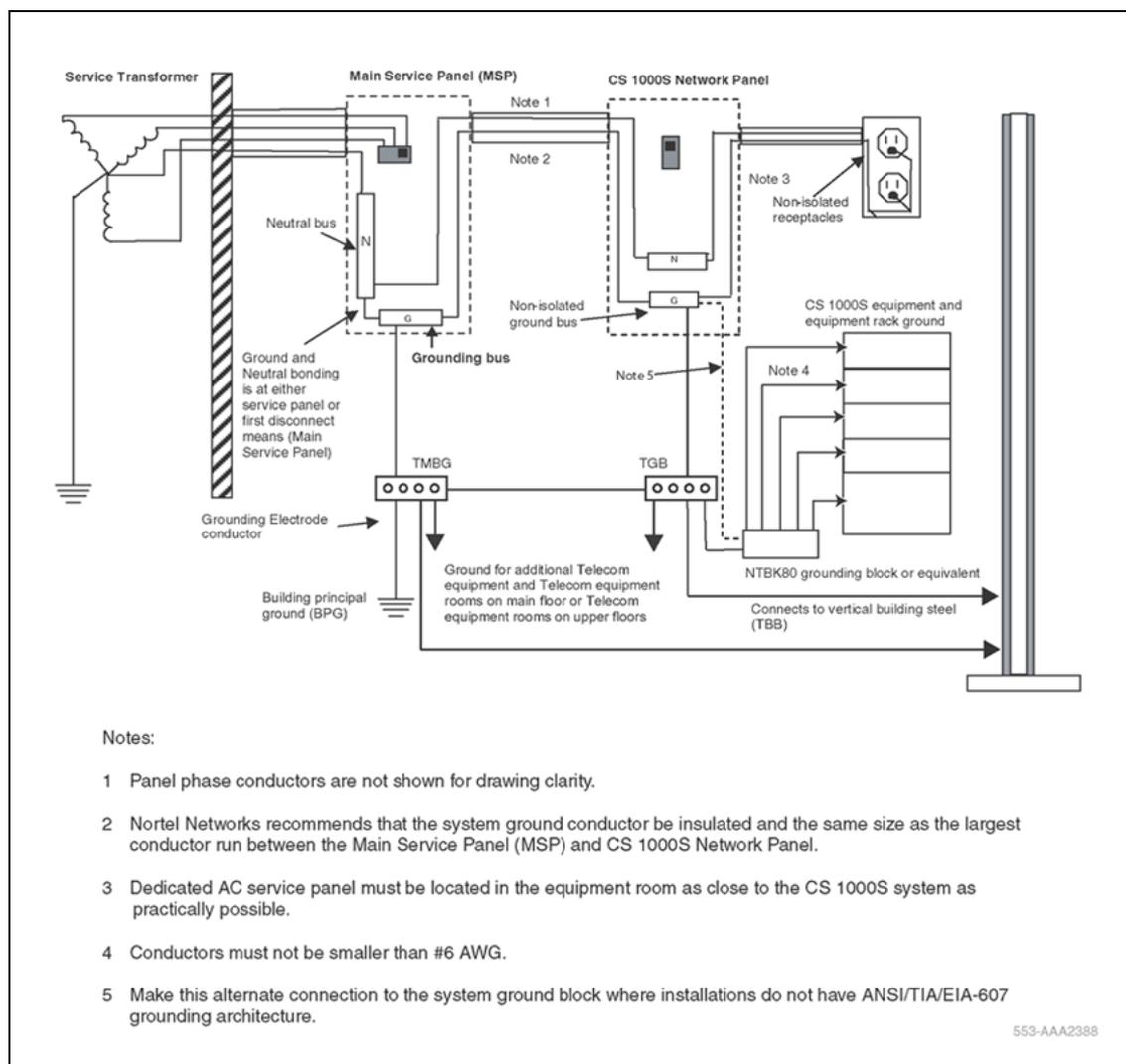
Power from each outlet must meet the input requirements of at least one Chassis or Cabinet system power supply, as listed in Table 7 "AC input requirements for each NTDK91 chassis and NTDK92 chassis expander (North America)" (page 49) through Table 10 "AC input requirements for each NTDK70 cabinet power supply" (page 50) beginning on Table 7

"AC input requirements for each NTDK91 chassis and NTDK92 chassis expander (North America)" (page 49). Check power requirements for other system equipment. Install additional outlets if required.

Table 7
AC input requirements for each NTDK91 chassis and NTDK92 chassis expander (North America)

Voltage	Recommended: 100-120 volts Maximum limits: 90 and 132 volts Single phase
Frequency	50-60 Hz
Power (I/P max)	550 VA maximum
Outlet Type	120 volt, 15 Amp supply

Table 8
AC input requirements for each NTDK91 chassis and NTDK92 chassis expander (Europe and UK)

Voltage	Recommended: 208/220 volts Maximum limits: 180 and 250 volts Single phase
Frequency	50-60 Hz
Power (I/P max)	550 VA maximum
Outlet Type	208/240 volt, 15 Amp supply
Note 1: As local power specifications vary, see a qualified local electrician when planning your power requirements. *Note 2:* The supplied power must be single-phase 240 or three-phase 208 Y, and must have a system ground conductor.	

Table 9
AC input requirements for each NTDK91 chassis and NTDK92 chassis expander (Germany)

Voltage	Recommended: 230 volts Maximum limits: 180 and 250 volts Single phase
Frequency	50 Hz
Power (I/P max)	550 VA maximum

Fuse	16A
Outlet Type	Receptacles by DIN regulation

Table 10
AC input requirements for each NTDK70 cabinet power supply

Voltage	Maximum rated input voltage 100-240 Volts RMS, single phase
Frequency	50-60 Hz
Power (I/P max)	750 VA minimum
Outlet Type	NEMA 5-15R for 120 Volt, 15 Amp supply NEMA 6-15R for 208/240 Volt, 15 Amp supply

Site requirements

The following is a list of required site features for an optimal AC-powered Small System installation.

If a dedicated panel cannot provide the conditions listed below, use an Isolation Transformer. Refer to .

- **Dedicated circuit breaker panel**

  A dedicated circuit breaker panel provides power only to the Small System and its related hardware, such as TTYs and printers.

  Note: You cannot always power a complete system from a single circuit-breaker panel. For example, when Expansion Chassis or Expansion Cabinets are located remotely.

- **Insulated copper ground conductor**

  The insulated copper ground conductor connects the ground bus in the dedicated panel to the main service panel ground or building ground reference. Route the insulated copper ground conductor through the same conduit as the supply conductors that feed the panel.

- **Outlets**

  Use a separate circuit for each device connected to the panel. Outlets that provide service to the chassis or cabinet must be close enough so that the power cord can reach the chassis or cabinet power supply.

  For systems equipped with Expansion Cabinets or Chassis, a separate outlet for each cabinet or chassis must be provided. Each outlet must be from separate circuits in the same panel.

Location of power outlets

The maximum distance between a power outlet and the system chassis or cabinet depends on the length of the power cord. In North America, the power cord is 9 ft 10 in. (3000 mm). In countries outside North America, the power cord is 8 ft 2 in. (2490 mm).

Alternative AC-powered installation

If a dedicated panel cannot provide optimal conditions, use an Isolation Transformer with the following characteristics:

- 120/208/240 V input, over-current protected at primary.

- 120/208/240 V available at secondary outputs, each circuit breaker-protected.

- Completely isolates primary and secondary windings from one another.

- Approved for use locally as a stand-alone user product (CSA, UL, or other locally recognized clear markings).

- Capable of providing power to all Small System equipment operating at the same time at full load.

- Equipment not related to the Small System must not be powered from a transformer that provides service to the Small System.

Uninterruptible Power Supply

For backup AC power, you can use an Uninterruptible Power Supply (UPS) to feed the Small System. The power requirement for a UPS is 550 VA per system. The maximum power requirement for a chassis and a chassis expander on the same breaker is 1100 VA. Install the UPS according to the manufacturer's instructions.

Isolation Transformer ground

The transformer ground must have the following characteristics:

- Separate grounds for primary and secondary windings, rather than a common ground.

- A "clean" and permanent SPG reference at the transformer secondary for the Small System.

Make sure that the ground conductors inside the transformer are sized correctly.

> *Note:* Do not ground the transformer or Small System to structural steel or water pipes. Connect them to a known building ground reference.

Receptacles

Receptacle requirements are as follows:

- When installed on the wall, install receptacles within reach of the chassis or cabinet power cords.

- The ground prong of each outlet must be connected by an insulated conductor to the system SPG.

If the transformer has an isolated secondary ground lug, use it as the SPG. If it does not, use the chassis ground lug of the transformer as the SPG.

Isolation Transformers
Transformers with pluggable power cords:

Step	Action
1	Connect the power cords of all Small System equipment to the outlets on the transformer secondary.
2	Secure an insulated conductor between the ground lug on the chassis or Main Cabinet of the Small System and the SPG lug on the transformer. Place a "DO NOT DISCONNECT" tag on it. Do not fasten or tie this conductor to the power cable feeding the Chassis system or cabinet power supply. *Note:* Power all equipment related with the Small System from the secondary of the transformer only. Ground all equipment to the secondary isolated ground lug. Do not connect equipment that is not related to the Small System to the Isolation Transformer that powers it.

—End—

Power the transformer primary through a dedicated circuit. If the primary has a pluggable cord, make an additional ground connection between the Small System SPG and a known building ground reference. This connection is very important for safe and reliable operation.

> **WARNING**
> Do not connect any system ground lines of the Small System to structural steel or water pipes, or other unreliable ground paths. Use a ground point known to be "clean" and permanent. Place a "DO NOT DISCONNECT" tag on it. Figure 6 "Typical pluggable cord Isolation Transformer wiring plan" (page 53) shows the pluggable cord connections.

Figure 6
Typical pluggable cord Isolation Transformer wiring plan

Notes:

1 Power source is site dependent. It may be from a shared panel or transformer. Wiring may vary accordingly. Wiring to panel must be housed in conduit.

2 Make SPG at the transformer secondary. If the transformer secondary has no isolated secondary ground lug, make SPG by tieing all system ground lines to the chassis lug on the transformer case. An insulated ground connection must be made between the SPG and a known building ground reference as per Note 3.

3 Terminate the Small System SPG insulated ground conductor as near as possible to the main building ground reference. Isolate the ground bus from panel housing if permitted by local codes. Conductor should be minimum AWG #6 (metric #40) at all points.

4 Wiring to receptacles must be in conduit unless they are mounted on the transformer case.

5 Connection may be made by metallic conduit. Additional copper conductor recommended.

6 Minimum AWG #6 (metric #40) insulated copper conductor connected to FGND lug on the Small System cabinet. Route separately from ac power cable.

7 Separate breaker required for each Small System cabinet. Breakers must be mounted on transformer if the receptacles are. If they are in a panel served by the transformer secondary, all connections between the receptacles and transformer must be in conduit.

8 Connect secondary neutral (XO) to system SPG.

9 Conduit required.

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Transformers without pluggable power cords   If the transformer does not have a pluggable cord, hard-wire the transformer to an electrical panel. Route all wires (including grounds) through a single conduit.

Some electrical codes permit the use of conduit as the only ground conductor between pieces of equipment.

Run a separate insulated ground conductor through the conduit to hold transformer chassis grounds together. Such a conductor maintains the safety ground connection in the event that the conduit becomes corroded or disconnected.

Run all ground lines through the same conduit as the phase conductors that serve the equipment. shows the Isolation Transformer connections.

Figure 7
Typical hardwired Isolation Transformer wiring plan

Notes:

1 Power source is site dependent. It may be from a shared panel or transformer. Wiring may vary accordingly. Wiring to panel must be housed in conduit.

2 Make SPG at the transformer secondary. If the transformer secondary has no isolated secondary ground lug, make SPG by tieing all system ground lines to the chassis lug on the transformer case. An insulated ground connection must be made between the SPG and a known building ground reference as per Note 3.

3 Terminate the small system SPG insulated ground conductor as near as possible to the main building ground reference. Isolate the ground bus from panel housing if permitted by local codes. Conductor should be minimum AWG #6 (metric #40) at all points.

4 Transformer primary wires must be in conduit. Wiring to receptacles must be in conduit unless they are mounted on the transformer case.

5 Connection may be made by metallic conduit. Additional copper conductor recommended.

6 Minimum AWG #6 (metric #40) insulated copper conductor connected to FGND lug on the Small System cabinet. Route separately from ac power cable.

7 Separate breaker required for each Small System cabinet. Breakers must be mounted on transformer if the receptacles are. If they are in a panel served by the transformer secondary, all connections between the receptacles and transformer must be in conduit.

8 Connect secondary neutral (XO) to system SPG.

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DC-powered version (Cabinet system only)

Each cabinet in a Cabinet system can be powered solely from a DC source if it is equipped with the following:

- NTDK72 DC power supply
- NTAK28 Junction Box

Table 11 "DC power requirements for each NTDK72 DC power supply" (page 55) lists the DC power requirements for the NTDK72 DC power supply.

Table 11
DC power requirements for each NTDK72 DC power supply

	Minimum	Nominal	Maximum
Input Range	-44 V DC	-52 V DC	-54 V DC
Noise (CMESS)	—	—	25 dBrnc
Current	—	—	12 Amps
AC Ripple	—	—	100 mv RMS

Note: The NTDK72 has a built-in circuit breaker that will trip if the voltage difference at its input terminals drops below -42.5 V DC + - 1.0 V DC.

> **WARNING**
>
> Do not allow the voltage difference between the input terminals of the NTDK72 to exceed 57 V DC. Doing so can result in damage to the equipment and a safety hazard to personnel.

The minimum size of the conductors required between the DC source and the Junction Box is shown in Table 12 "Recommended wire size" (page 55).

Table 12
Recommended wire size

Size (AWG)	Size (Metric)
6	#40
8	#35
10	#25

Connect these components together as shown in Figure 8 "DC power supply connections" (page 56). Make sure the Main Cabinet ground post is connected to the building ground reference by a minimum AWG #6 (metric #40) insulated conductor. Connect the input terminals of the NTAK28

Junction Box to a clean DC power source meeting the requirements shown in Table 11 "DC power requirements for each NTDK72 DC power supply" (page 55).

Figure 8
DC power supply connections
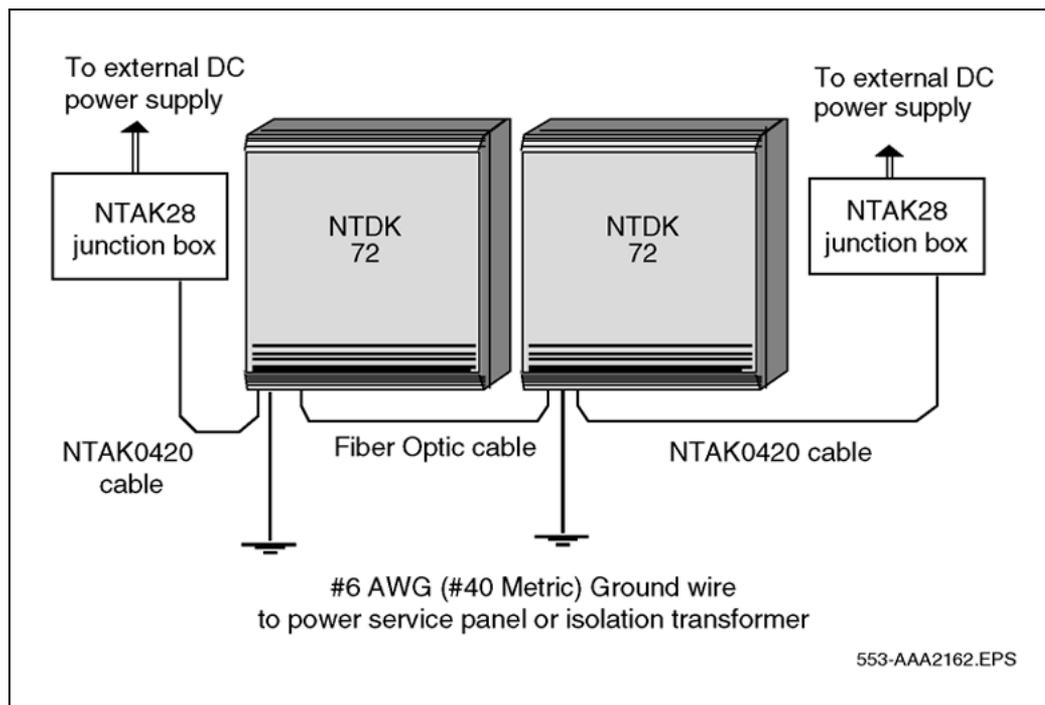
Power consumption

Table 13 "Circuit card and daughterboard power consumption" (page 56) provides the circuit card power consumption. Use the worksheets provided (see "Worksheet 7: Cabinet system power consumption" (page 309) and "Worksheet 8: Chassis system power consumption" (page 312)) to determine the power consumption for the system.

Table 13
Circuit card and daughterboard power consumption

Circuit card	Type	% active sets (off-hook)	Power consumption (watts)
Mail	Meridian Mail	steady state	35
NT1R20	Off premise Station analog line card	50%	22
NT5K02	Flexible analog line card	50	26
NT8D02	Digital line card	100%	25
NT8D09	Message-waiting line card	50%	26

Circuit card	Type	% active sets (off-hook)	Power consumption (watts)
NT8D14	Universal trunk card	DID-enabled	28
NT8D15	E&M trunk card	N/A	29
NT8D16	Digitone receiver card	N/A	6
NTAK02	SDI/DCH card	N/A	10
NTAK10	2.0Mb DTI card	N/A	12
NTAK79	2.0Mb PRI card	N/A	12
NTBK22	MISP card	N/A	12
NTBK50	2.0Mb PRI card	N/A	12
NTDK16	48-port Digital Line Card (chassis only)	100%	75
NTDK20	SSC card	N/A	15
NTRB21	1.5Mb DTI or PRI card	N/A	12
NTVQ01	Voice Gateway Media Card (32-port)	N/A	18

Auxiliary equipment power

Terminals, printers, modems, and other data units used with Small Systems require special wiring considerations.

Power for system equipment in the switch room must:

- be powered from the same panel or transformer as the Small System
- be grounded to the same panel or transformer as the Small System
- be labeled at the panel to prevent interruption that is not authorized
- not be controlled by a switch between the breaker and the equipment

Service receptacles for AC-powered Small Systems and related equipment must be:

- rated for 120 or 240 V, 15 or 20A, 50-60 Hz, 3-pole, 3-wire, grounded
- grounded to the same location so as to form a SPG

Modem requirements

Equip the system with a modem to allow remote access. Refer to the section on modem setup requirements in *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for information about setting up the modems recommended for use with Small Systems.

Note: In the UK, British Telecom RACE modems require a Modem Eliminator (NULL Modem without hardware handshaking) A0378652 F-F converter or A0381016 M-F converter.

With or without Meridian Mail
The minimum requirement is a 1200 bps auto-answer modem.

If an error-correcting modem is connected to the Small System, all flow-control and error-correcting functionality of the modem must be disabled to ensure correct operation. Refer to the modem manufacturer's instructions for information.

Customer Configuration Backup and Restore
The Customer Configuration Backup and Restore (CCBR) feature provides the ability to store the configuration database of the Small System on an external hard drive of an IBM-type PC or Macintosh-type computer.

The CCBR feature can be invoked on-site with the use of a modem eliminator, or remotely over a modem connection.

Communications software
Communications software compatible with XModem CRC protocol is required to operate the CCBR feature. This requirement applies to on-site and remote access.

On-site access
On-site access to the Small System can be made by directly connecting a computer to SDI port 0, 1, or 2.

Note: Connect a modem eliminator between the SDI cable and the computer cable for on-site computer access.

Remote access
Remote access to the Small System is established by connecting SDI port 0, 1, or 2 on the SSC card to an analog line (Central Office line) through an on-site modem. This allows the computer to dial directly into the system from a remote location.

Detailed information about the CCBR feature can be found in *Communication Server 1000M and Meridian 1 Small System Maintenance (NN43011-700)*.

Maintenance and administration terminals
Refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for information about setting up terminals recommended for use with a Small System.

Under some conditions, a Modem Eliminator (NULL Modem without hardware handshaking) A0601397 F-F converter or A0601396 M-F to interface the TTY to the system is required.

The following describes the minimum requirements for a TTY device.

Without Meridian Mail
When the system does not have Meridian Mail installed, and it will not have Meridian Mail installed in the future, the minimum requirement is a VT100 compatible device.

With Meridian Mail
With Meridian Mail installed, use a VT220-compatible device.

On-site access
Equip each system with an M2616 or M2008 telephone with display. These telephones act as maintenance telephones.

You can use many different TTY terminals to access the Small System. However, a VT220 terminal is recommended as an on-site terminal. You can use this terminal to perform service changes, maintenance and diagnostic functions, and CallPilot administration activities.

Remote access
Although you can use several types of modems to access the system, a 2400 baud auto-answer modem is the recommended modem. A 1200 baud modem is the minimum requirement. You can use the modem to perform service changes, maintenance and diagnostic functions, and CallPilot administration activities from a remote location.

Note: You can perform additional maintenance functions through remote access on a Small System. For additional information, refer to *Communication Server 1000M and Meridian 1 Small System Maintenance (NN43011-700).*

Optivity Telephony Manager (OTM)
The Small System supports the OTM application. For information about OTM requirements, refer to *Telephony Manager 3.1 System Administration (NN43050-601)* or *Telephony Manager 3.1 Telemanagement Applications (NN43050-602).*

Cross-connect terminal requirements
Allow for future expansion and equipment changes at the cross-connect terminal.

The cross-connect terminal must have enough space for connecting blocks to terminate the following wires:

- ten 25-pair cables from each cabinet (the Main Cabinet and, if equipped, each Expansion Cabinet)

- six conductors comprising the AUX cable from the Main Cabinet

- five 25-pair cables from each chassis

- four 25-pair cables from the chassis expander

- four conductors for the AUX cable from the chassis

- one 25-pair cable from each QUA6 Power Failure Transfer Unit (PFTU)

- wiring from telephone sets and trunks

The BIX cross-connect system is recommended for use with the Small System. However, use of this system is not mandatory. You can use some other cross-connect systems (for example, the Krone Test Jack Frame for the UK and the Reichle Masari cross-connect terminal for Germany).

Only allow authorized personnel to access the Krone Test Jack Frame. Install the Krone Test Jack Frame in a locked room or in an environment that prevents free access to the equipment. The Krone Test Jack Frame must meet this safety requirement to receive approval.

You can find information about the BIX cross-connect system in the following documents:

- *BIX In-Building Cross-Connect System Material Description (631-4511-100)*

- *BIX In-Building Cross-Connect System Material Installation and Servicing (631-4511-200)*

Refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for additional information about the BIX, Krone Test Jack Frame, and Reichle Masari cross-connect terminals.

IP installation requirements for Small Systems

Connectors for IP cabinets or chassis equipped with 100BaseF or 100BaseT daughterboards must be identified in advance in order to secure the proper cabling cross-connect. These connectors are the responsibility of the customer.

100BaseF daughterboards

The 100BaseF Expansion Cabinets or Chassis can be located up to 2 km (1.25 mi) from the Main Cabinet/Chassis or customer-supplied LAN equipment.

100BaseT daughterboards

IP Expansion Cabinets or Chassis equipped with 100BaseT or 100BaseF daughterboards can be located up to 100 m (300 ft) from the Main Cabinet, Chassis, or customer-supplied LAN equipment.

Note: For IP connections greater than the 100BaseT or F solutions, third-party media convertors are required.

Power supplies

Contents

This section contains information on the following topics:

Introduction

This chapter describes the Cabinet system AC power supply (NTDK70), DC power supply (NTDK72), and reserve power requirements, and the operation of the Power Failure Transfer Unit (PFTU). The Chassis system NTDK15 power supply is described on "Chassis system power supply features" (page 69).

Features of the Cabinet system power supply

Dimensions and weight

The AC/DC and DC power supplies measure approximately 12.5 inches (305 mm) high, 5 inches (127 mm) wide, and 10 inches (245 mm) deep.

The AC power supply weighs approximately 12 lb (5.5 kg), while the DC power supply weighs approximately 8 lb (3.5 kg).

AC/DC power supply features

The NTDK70 AC power supply has the following features:

- A current-limiting circuit, which limits the surge of current on the input line when the system is first switched on.

• Accommodates a reserve power system. The system continues to operate on DC reserve power in case of AC power failure.

> *Note:* The NTDK70 AC power supply cannot power up on battery alone. If the NTDK70 is powered down while operating on DC reserve power, then AC power is required to power up the system.

• Battery charging for the reserve power system. Charging current in a worst-case scenario (when Call Pilot is installed) is 1.0 amp.

• Power (± 15V) for one attendant console.

• Generation of a system line transfer signal and power (-52V) for the PFTU (250 MA maximum).

• Differential mode and common mode EMI filtering of input.

• Input power (-52VDC) for the Call Pilot power supply (NTAK13).

DC power supply features

The NTDK72 DC power supply has the following features:

• Power (± 15V) for one attendant console.

• Generation of a system line transfer signal and power (-52V) for the PFTU (250 MA maximum).

Voltage

The AC/DC power supply and the DC power supply provide +5.1, +8.5, +15, -15V, -150V, -52V power supplies and filtered -48V.

There is a 1.0-second start-up delay on the +5V rail.

Ringing generator

The AC/DC power supply and the DC power supply provide the ringing generator for telephones:

• Ringing voltage: 70, 75, 80, 86V.

• Ringing frequency: 20, 25, 50 Hz, switch selectable.

• Ring sync: A pulse 500 us wide, 6 or 11 ms (±3 ms) before the positive going zero crossing of the ringing waveform (11 ms for 20/25 Hz).

• Power: The output capability is 8VA which is capable of ringing 8CA4 ringers.

Power supply LED

The LED on the power supply faceplate labeled "DC" is turned off whenever there is a problem with the power supply.

Under-voltage

Under-voltage to the AC/DC or DC power supply results in partial failure of the system. The faceplate LED labeled "DC" is turned off.

> **WARNING**
> Under-voltage, in the case of +5.1V, results in the complete shutdown of the system.

Table 14 "Nominal and under-voltage limits of NTDK70 and NTDK72 power supplies" (page 65) outlines the nominal and under-voltage limits of the power supply.

Table 14
Nominal and under-voltage limits of NTDK70 and NTDK72 power supplies

Nominal	Under-voltage limit	Power supply status
+5.1V	+3.8V	Complete Shutdown
8.5V	+6.4V	Partial failure
-150V	-100.0V	Partial failure
+15V	+11.2V	Partial failure
-15V	-11.2V	Partial failure
-48V	-36.0V	Partial failure
Ring (Pk V)	70V	Partial failure
-52V	-45V	Partial failure

Over-voltage

An OVP (Over-Voltage Protection) circuit will shut down the power supply if the output voltage exceeds the limits given in Table 15 "Nominal and over-voltage limits of NTDK70 and NTDK72 power supplies" (page 65).

Table 15
Nominal and over-voltage limits of NTDK70 and NTDK72 power supplies

Nominal voltage	Overvoltage limit	Power supply status
+5.1V	+6.4V	Complete Shutdown
+8.5V	+10.6V	Complete Shutdown
-150V	-187.5V	Complete Shutdown
+15V	+18.7V	Complete Shutdown
-15V	-18.7V	Complete Shutdown
-48V	N/A	N/A

Nominal voltage	Overvoltage limit	Power supply status
Ring (Pk V)	150V	Complete Shutdown
-52V	-58V	Complete Shutdown

All outputs in a shutdown state are reset by the Small System Controller (SSC) card.

The system power does not automatically reset when there is over-voltage on the -52V DC output. Manual intervention is required. The manual int button is located on the faceplate of the SSC card.

Temperature sensor

The power supplies are sensitive to the temperature of the cabinet and the system power. A thermostat is located at the top of the power supply unit. The AC or DC input breaker is tripped for temperatures higher than 80°C (176°F).

Reserve power LED

The NTDK70 AC power supplies oversee the status of the reserve power system. When the breaker on the NTAK28, NTAK75, or NTAK76 breaker assembly trips, the "Batt" LED on the NTDK70 faceplate is turned off.

PFTU operation

Power is switched over to the PFTU during any of the following conditions:

- The CPU sends a signal to the PFTU.

- A power failure occurs.

- A CPU failure occurs.

- The PFTU is manually activated.

- The fiber link to an Expansion Cabinet fails (PFTU for that cabinet only).

The Cabinet system power supply connects to the PFTU through the AUX connector at the bottom of the Main Cabinet, and in each Expansion Cabinet. Table 16 "Auxiliary cable pinouts" (page 66) provides the pinouts at the cross-connect terminal for the Auxiliary cable.

Table 16
Auxiliary cable pinouts

Cable	Signal
BL-W 1 Dot	BRTN
BL-W 2 Dot	BRTN
O-W 1 Dot	-48 V AUX

Nortel Networks Confidential

Cable	Signal
O-W 2 Dot	PFTS
G-W 1 Dot	-15V AUX
G-W 2 Dot	+15V AUX
BR-W 1 Dot	-
BR-W 2 Dot	-

Reserve power

Discharge requirements

Reserve batteries must be able to provide 500 watts of power to each cabinet. This is a worst-case figure based on the maximum power consumption per cabinet.

Backup options

The options available when backing up the AC-powered Cabinet system are as follows:

- Use customer-supplied batteries along with the NTAK28 breaker assembly.

- Connect an Uninterrupted Power Supply (UPS) to the system.

- Use Nortel-supplied NTAK75 or NTAK76 battery units.

> **WARNING**
> Always follow the manufacturer's instructions when installing batteries.

Customer-supplied reserve batteries with NTAK28

Customer-supplied batteries can be used as long they meet the requirements listed in Table 17 "Reserve battery requirements" (page 68). One NTAK28 breaker assembly is required per cabinet.

NTAK75 or NTAK76 battery units

Two battery units are available. The NTAK75 supplies a minimum of two hours backup at full load, while the NTAK76 supplies a minimum of fifteen minutes backup at full load.

Table 17
Reserve battery requirements

Sealed cells	Cell float voltage	String float voltage
23	2.30 – 2.36	52.95 – 54.25
24	2.20 – 2.26	52.95 – 54.25

Uninterrupted Power Supply (UPS)

A 750VA Uninterrupted Power Supply (UPS) may be connected to AC-powered systems in order to provide a continuous supply of AC power.

If two cabinets are equipped, two 750VA UPSs or one 1.5KVA UPS can be used.

Battery charging in AC-powered systems

During normal operation, the AC power supply (NTDK70) provides a constant float voltage to the reserve batteries. This charger voltage is not adjustable and will not provide equalization voltages. See Table 18 "NTDK70 or AC power interface to reserve power systems" (page 68).

Table 18
NTDK70 or AC power interface to reserve power systems

	Minimum	Nominal	Maximum
Float Voltage	52.95 Volts	53.6 Volts	54.50 Volts
Charge Current	1.0 Amps	—	7.0 Amps
Note: The charge current available to the reserve batteries depends on the system configuration and the line size.			

Reserve time

Table 19 "Reserve time" (page 69) outlines the Ampere hours required (AHR) per cabinet during a power failure. The reserve times are based on nominal load for a typical installation.

Table 19
Reserve time

Duration of Power Failure	AHRs required per cabinet
30 − 40 minutes	6 AHR
1.5 − 2 hours	12 AHR
3 − 4 hours	25 AHR

Chassis system power supply features

This section describes the Chassis system NTDK15 AC power supply.

Dimensions and weight

The Chassis AC power supply is factory-installed in the chassis and is not accessible. The power supply measures approximately 1.75 in. (44 mm) high, 8 in. (203 mm) wide and 10 in. (254 mm) deep, and weighs approximately 3 lb (1.4 kg).

AC power supply features

The Chassis AC power supply has the following features:

- A current limiting circuit which limits the surge of current on the input line when the system is first switched on.

- All outputs fully regulated.

- Universal 100-240 VAC input.

- 363 Watt total output power.

- Meets CISPR B emission per EN 55022.

- Power status indicator LED is located on the top front left corner of the chassis.

    The green LED indicates all voltages are within specification. The LED is off when one or more voltages are not within specification.

- Ringing voltage: 70, 75, 80, or 86 Vrms depending on DIP switch settings.

- Ringing frequency: 20, 25, or 50 Hz depending on DIP switch settings.

    Note: The DIP switch discussed here is located on the front top plate of the chassis, and can only be accessed with the chassis faceplate removed.

- Cooling is provided by a fan mounted inside the chassis.

- Power: The output capability is 5VA, which is capable of ringing 5C4A ringers.

- Provides ring synchronization (zero current crossing) signal.

- Power on/off switch.

- Power status output to CPU.

Voltage

The chassis AC power supply provides +5.1, +8, +15, -15, and -48V. -120V/-150V is selected or disabled by DIP switch settings.

There is a 1.0-second start-up delay on the +5V rail.

Over-voltage

An OVP (Over-Voltage Protection) circuit shuts down all outputs if the +5 V output voltage exceeds the over-voltage threshold.

Under-voltage

An under-voltage protection circuit shuts down all outputs if +5V output is below the under-voltage threshold.

There is a 1.0-minute recovery delay from an under-voltage condition.

Developing an equipment layout plan for Cabinet systems

Contents

This section contains information on the following topics:

Introduction

Before installing the Small System, you need to develop an equipment layout plan to determine where each system component will be positioned.

Consideration should be given to the lengths of the various cables in order to make the best use of space available. Refer to Table 12 "Recommended wire size" (page 55) for a complete description of Small System cable and wire specifications.

Preparation of the site according to the plan is critical. Site preparation consists of making sure the site is ready to accept the equipment and that items such as power outlets and backboards are correctly installed.

General layout guidelines

> **DANGER**
> The mounting surface must be able to support at least 100 lb (45 kg). For wall-mounted systems, Nortel recommends that you secure a backboard consisting of 3/4-in. (20-mm) plywood, or other similar material, to the surface of the wall to hold the equipment.

Follow the guidelines below to assist you in positioning the system equipment. If you plan on installing one or more Expansion Cabinets, read the section called "Additional considerations for multiple-cabinet systems" (page 73).

- The recommended method of system cabinet installation is wall-mounting. If you cannot mount the cabinets on the wall (for example, if there is not enough wall space), you can mount each cabinet on an optional pedestal. However, you still need wall space for installing a cross-connect terminal and other optional equipment.

- Each NTAK11 cabinet measures 25 in. (635 mm) high by 22 in. (560 mm) wide by 12 in. (305 mm) deep, or 14 in. (356 mm) deep with the newer cabinet door.

- Leave adequate space for one or more Expansion Cabinets. When possible, mount the Expansion Cabinets next to each other horizontally (horizontal expansion) to ensure proper heat dissipation.

- If horizontal expansion is not possible, vertical expansion is permitted for two cabinets. Make sure the Expansion Cabinet is mounted above the Main Cabinet.

  Nortel does not recommend vertical expansion of three or more cabinets. Such a configuration makes reaching the topmost cabinet difficult.

  Note: Temperature limits are more stringent when expanding vertically. Review the temperature limits stated in "Environmental requirements" (page 38) of this document before deciding to expand vertically. Do not install an Expansion Cabinet on top of an existing floor-mounted cabinet.

- When planning for a system that is equipped with DTI/PRI capability, allow space on the backboard for the channel service unit (CSU).

- Leave at least 6 in. (155 mm) above the mounting bracket and any obstruction (such as a pipe or conduit) so that there is room to lift the cabinet on and off the bracket.

- Leave at least 12 in. (305 mm) between the top of a cabinet and the ceiling to ensure proper ventilation.

- Leave 10 in. (255 mm) between the bottom of the lower cabinet and the floor to prevent water damage and to allow for convectional cooling.

- Do not place the cross-connect terminal above a cabinet. Debris from the cross-connect terminal may drop into the cabinet through the top ventilation slots and cause damage.

- Allow adequate space for the battery backup unit, accounting for the cable-length limitation as determined by the choice of a wall-mounted or floor-mounted battery back-up unit.

- If the NT1R20 Off-Premise Station card is used, allow for proper installation (according to local practices).

- Ensure power outlets are within reach of each system cabinet. For cable and wire specifications, refer to Table 12 "Recommended wire size" (page 55) and "Expansion cabling" (page 108).

Additional considerations for multiple-cabinet systems

For multi-cabinet systems the following guidelines apply for both horizontal and vertical expansion:

- The maximum distance between the Main Cabinet and each Expansion Cabinet is 1.8 mi (3 km).

- The minimum distance between cabinets when mounted above one another (vertical expansion) is 12 in. (305 mm).

- The minimum distance between cabinets when mounted next to each other (horizontal expansion) is defined by an alignment bracket as shown in Figure 11 "Mounting bracket position" (page 76). However, this is the minimum distance; the cabinets can be positioned further apart to suit site requirements.

    Note: The equipment layout plans shown in this chapter are applicable to fiber-optic connected cabinets installed within close proximity to each other (such as on the same wall). These layout guidelines are not as stringent if the cabinets are located in separate rooms, on different floors, or in different buildings.

Systems using NTAK75 or NTAK76 reserve power

The mounting location of either the NTAK75 or the NTAK76 backup unit is governed by the location of the cabinets and the length of the NTAK0410 cable. The NTAK0410 cable is 6 ft (1830 mm) long.

Equipment layout plan for wall mounting

Figure 9 "Typical minimum distance layout of wall-mounted cabinets (horizontal expansion)" (page 74), Figure 10 "Typical minimum distance layout of wall-mounted cabinets (vertical expansion)" (page 75), and Figure 12 "Typical layout of floor-mounted cabinets" (page 77) show typical wall layouts. Figure 10 "Typical minimum distance layout of wall-mounted cabinets (vertical expansion)" (page 75) and Figure 12 "Typical layout of floor-mounted cabinets" (page 77) use BIX cross-connection equipment. Use of other types of terminal blocks and equipment alters the layout. As a result, you may need to adjust the height at which you place the cabinets in relation to other equipment. You may also need to adjust the distances the power outlets are from the backboard on AC-powered systems. The positions for the mounting brackets are shown in Figure 11 "Mounting bracket position" (page 76).

Figure 9
Typical minimum distance layout of wall-mounted cabinets (horizontal expansion)
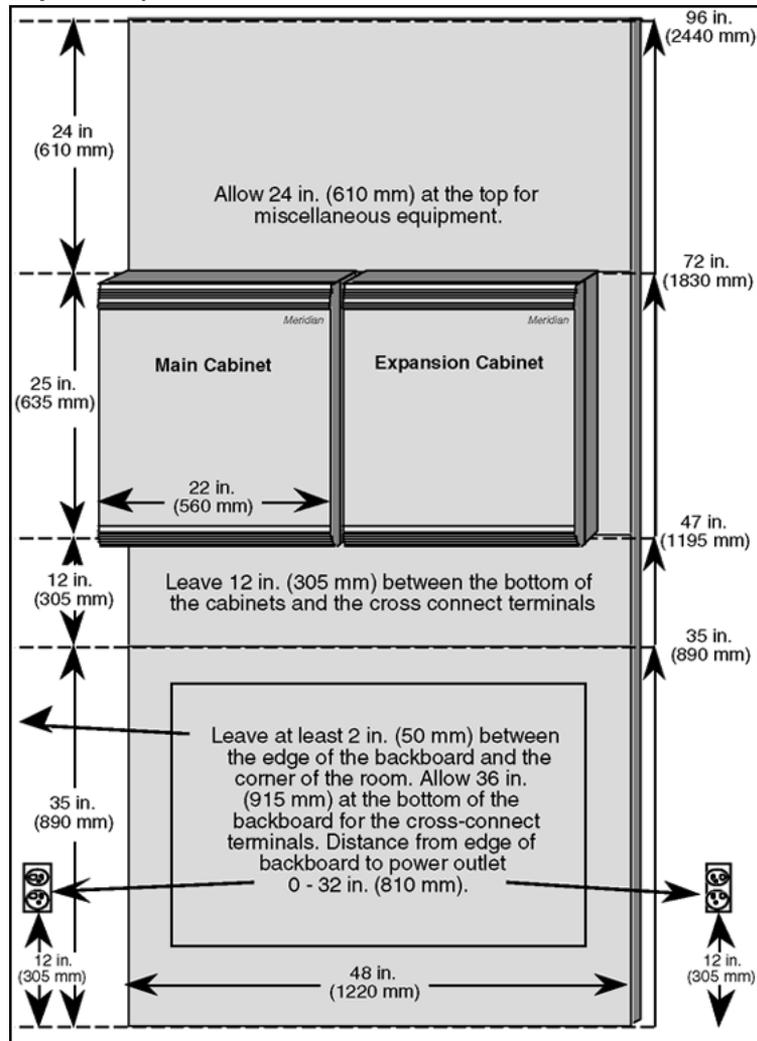
553-8326.EPS

Figure 10
Typical minimum distance layout of wall-mounted cabinets (vertical expansion)
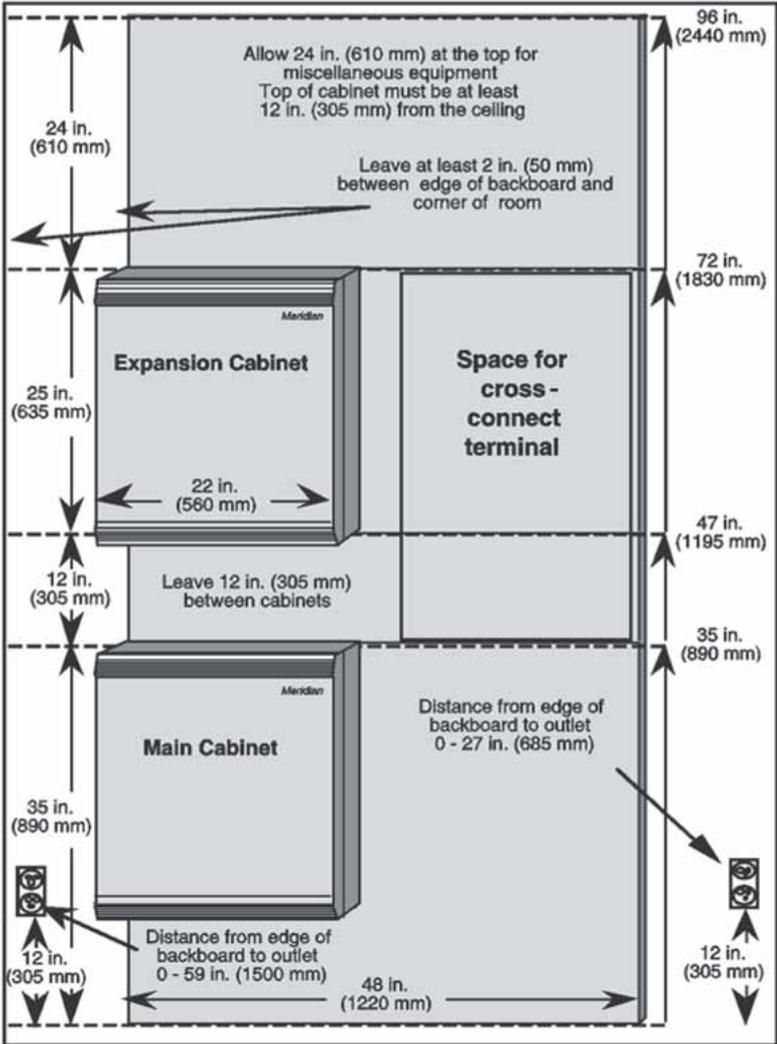
553-AAA1089

Figure 11
Mounting bracket position

48 in.
(1220 mm)

11 in. (280 mm) to
center of bracket

Main Cabinet
(horizontal expansion)
or
Expansion Cabinet
(vertical expansion)

Alignment
bracket

Expansion Cabinet
(horizontal expansion)

68.5 in.
(1740 mm)

37 in.
(940 mm)

Position cabinet
according to layout plan

Expansion Cabinet
(vertical expansion)

31.5 in.
(800 mm)

553-8328.EPS
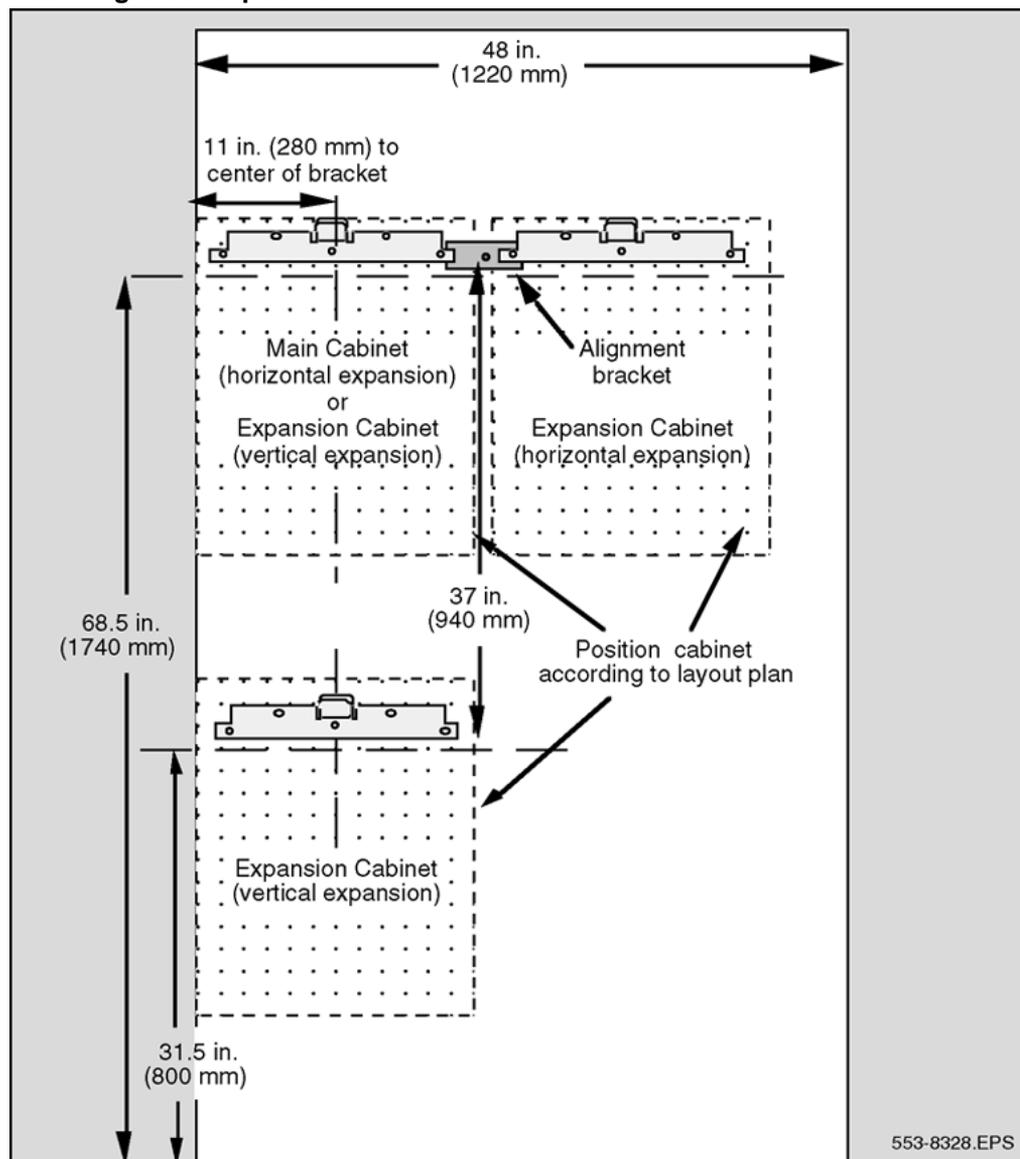
Equipment layout plan for floor mounting

An optional cabinet pedestal is used for floor-mounting when it is not possible to mount the cabinets on a wall.

The available floor space must be large enough to accommodate the Main Cabinet and one or more Expansion Cabinets, as shown in Figure 12 "Typical layout of floor-mounted cabinets" (page 77).

Note: Although you can be installing only a Main Cabinet at this time, leave enough space for Expansion Cabinets to avoid problems in the future.

Wall space must be available for the cross-connect terminal, the cross-connect cables, the NTAK75 or NTAK76 battery unit if required, and any miscellaneous equipment (such as a power supply for digit displays on telephones).

⚠️	**WARNING** Make sure that cabinet placement does not allow debris from sources such as cross-connect terminal activities to fall into the ventilation slots located at the top of the cabinet. Leave at least 12 in. (305 mm) of space between the top of the cabinet and any obstruction (such as a shelf) to permit adequate air circulation.

Figure 12
Typical layout of floor-mounted cabinets
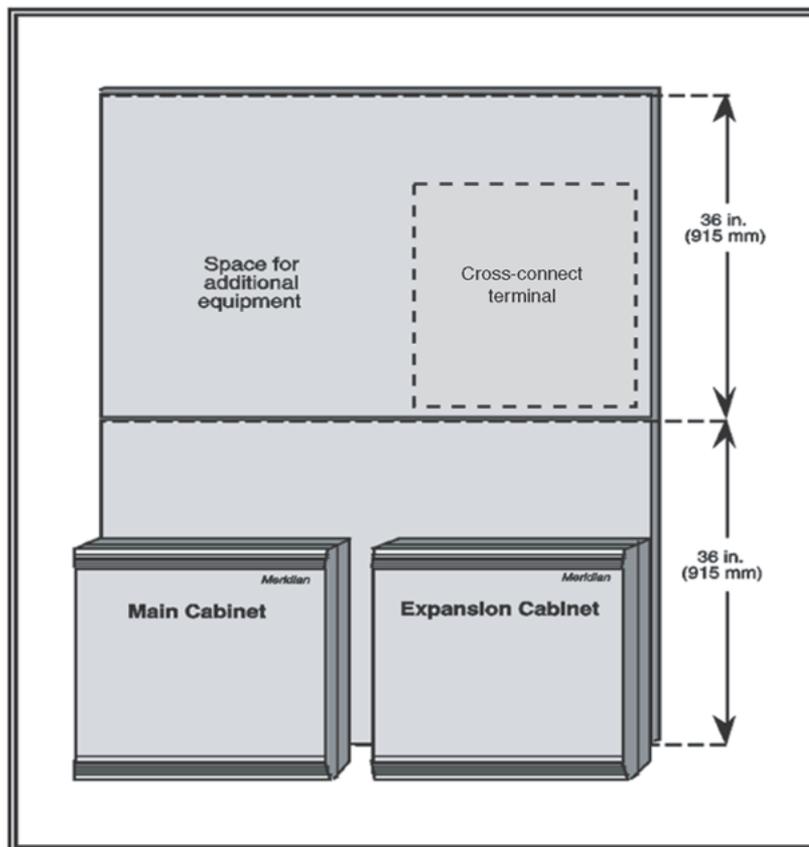
553-AAA1090

Reserve power supply layout and installation planning

The mounting location of either the NTAK75 or the NTAK76 reserve power unit is governed by the location of the Main and Expansion Cabinets, and the length of the NTAK0410 cable (the NTAK0410 cable is 6 ft [1830 mm] in length). Figure 13 "Typical placement of NTAK76 (horizontal cabinet expansion)" (page 78) through Figure 18 "Typical placement of NTAK75 (three-cabinet system)" (page 83), beginning on Figure 13 "Typical placement of NTAK76 (horizontal cabinet expansion)" (page 78), show typical placement of the NTAK75 and NTAK76 for horizontal cabinet expansion, vertical cabinet expansion, and three-cabinet systems.

Figure 13
Typical placement of NTAK76 (horizontal cabinet expansion)
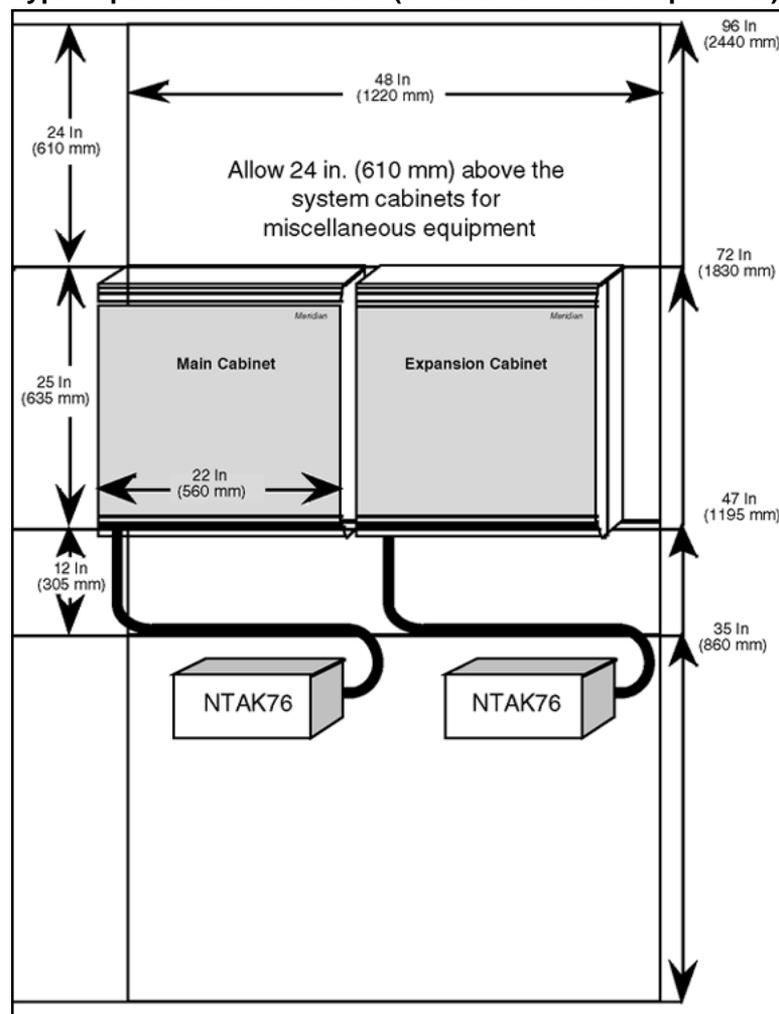
The center line of the NTAK76 can be placed a maximum of 2 ft (610 mm) to the right and 4 ft (1220 mm) to the left of the cabinet center line. These distances are based on the top of the NTAK76 being positioned 1.5 ft (460 mm) below the bottom of the cabinet.

Figure 14
Typical placement of NTAK75 (horizontal cabinet expansion)
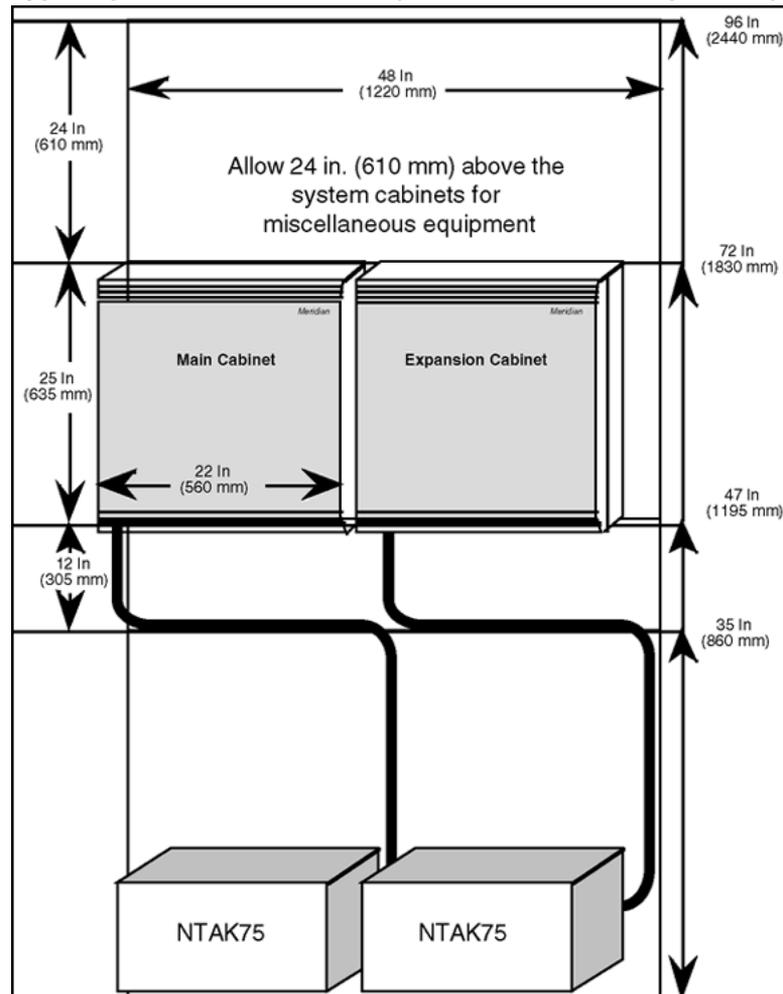
553-8331.EPS

The center line of the NTAK75 can be placed a maximum of 0.5 ft (152 mm) to the right and 2.5 ft (760 mm) to the left of the cabinet center line. These distances are based on the cabinets being mounted at the recommended mounting heights, shown in Figure 14 "Typical placement of NTAK75 (horizontal cabinet expansion)" (page 79), for the horizontal mounting configuration.

Figure 15
Typical placement of NTAK76 (vertical cabinet expansion)
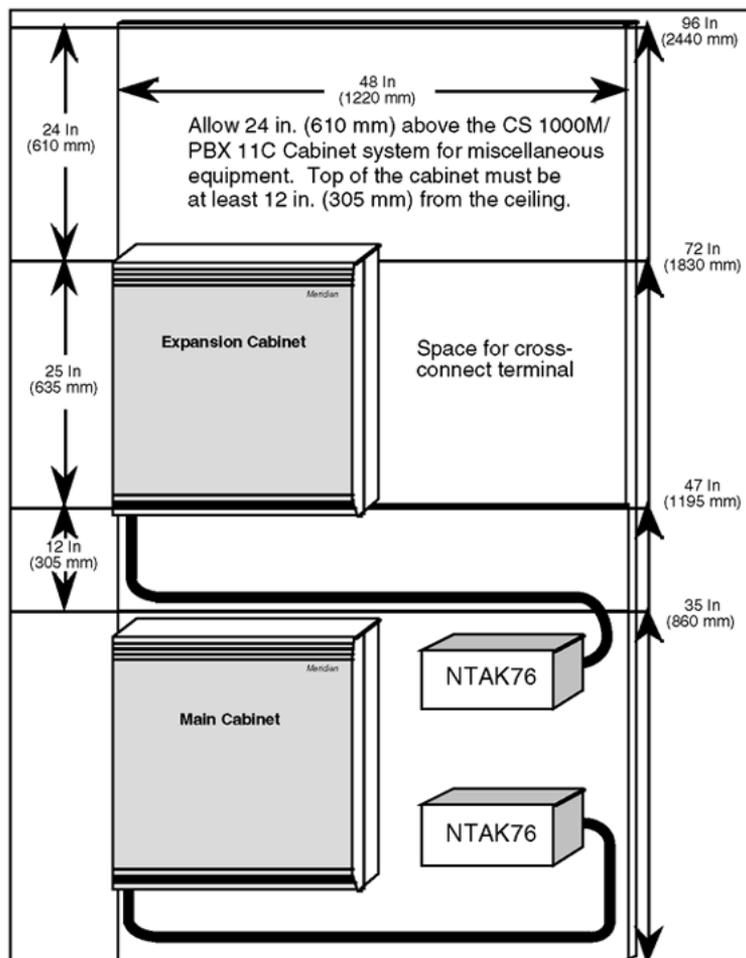
The center line of the NTAK76 can be placed a maximum of 2 ft (610 mm) to the right and 4 ft (1220 mm) to the left of the cabinet center line. These distances are based on the top of the upper NTAK76 being positioned 1.5 ft (460 mm) below the bottom of the Expansion Cabinet, and the bottom of lower NTAK76 being positioned 1.5 ft (460 mm) below the bottom of the Main Cabinet.

Figure 16
Typical placement of NTAK76 (three-cabinet system)
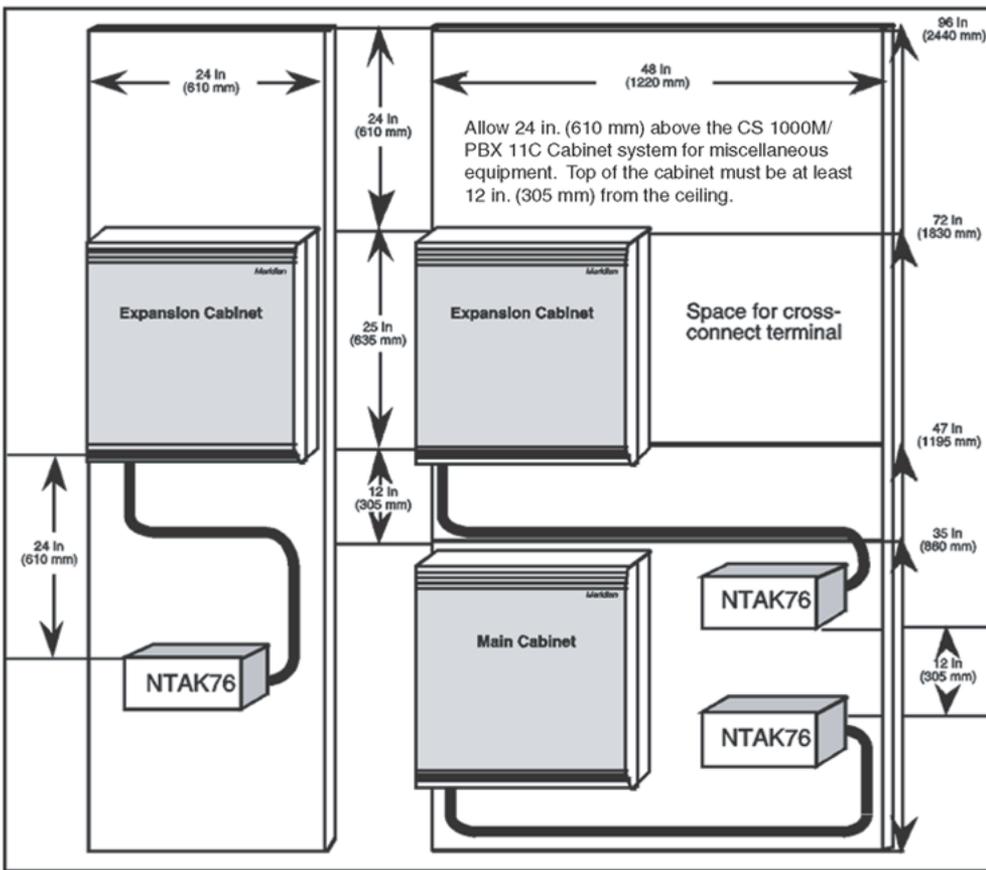
553-AAA2166.EPS

Figure 17
Typical placement of NTAK75 (vertical cabinet expansion)
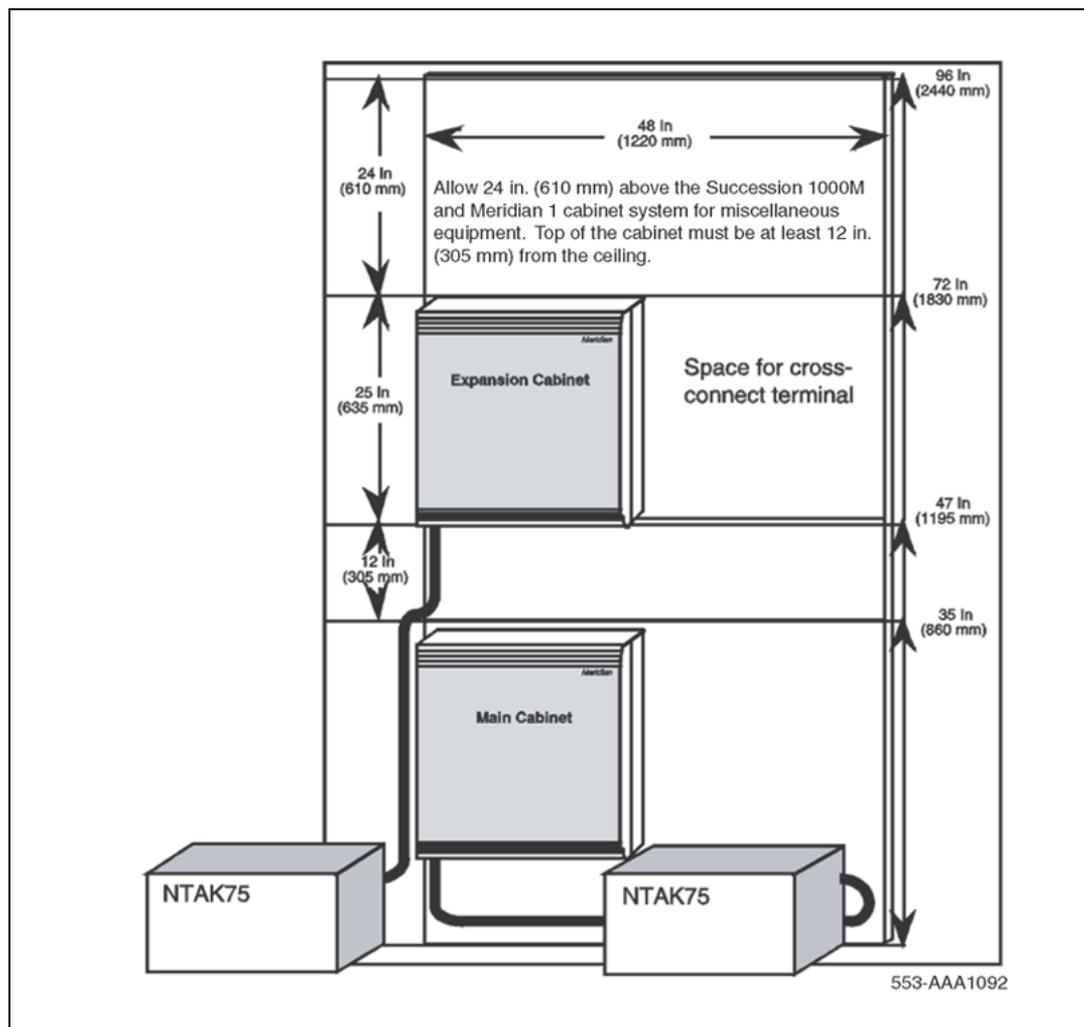
The center line of the NTAK75 for the Expansion Cabinet can be placed a maximum of 2.5 ft (760 mm) to the left of the center line of the cabinet. The center line of the NTAK75 for the Main Cabinet can be placed a maximum of 2 ft (610 mm) to the right of the center line of the cabinet. These distances are based on the cabinets being mounted at the recommended heights, as shown in Figure 17 "Typical placement of NTAK75 (vertical cabinet expansion)" (page 82), for the vertical mounting configuration.

Figure 18
Typical placement of NTAK75 (three-cabinet system)
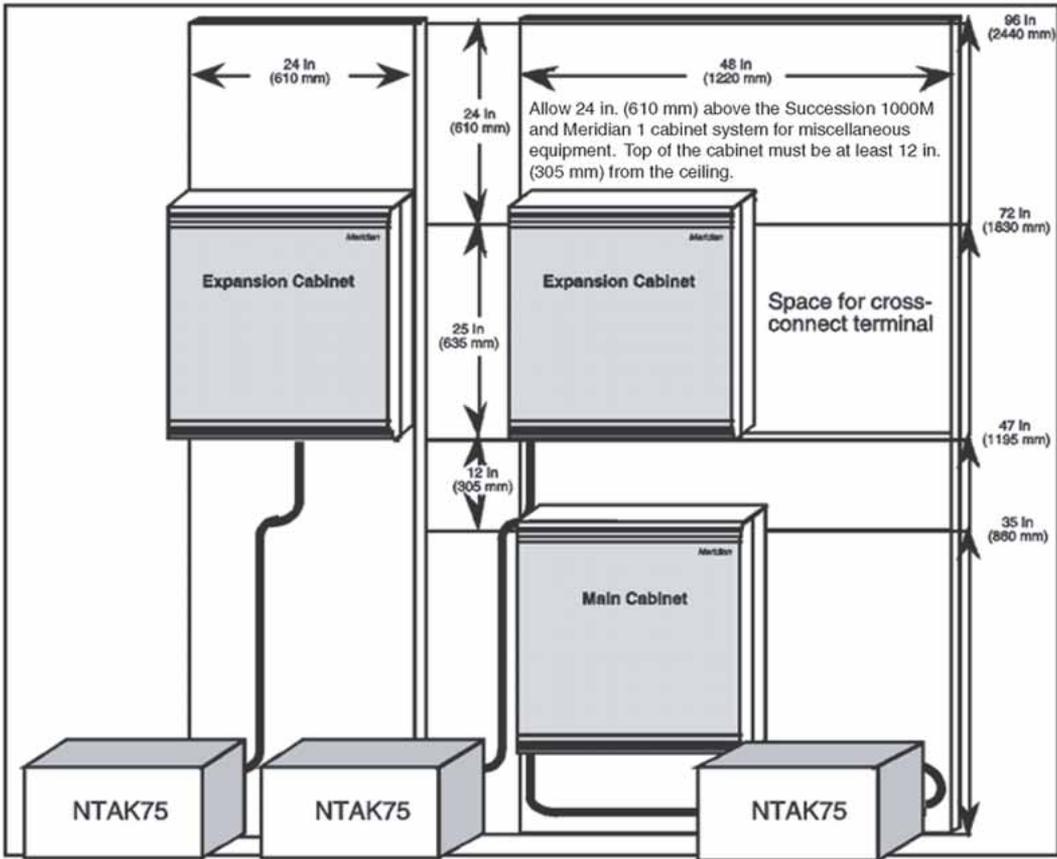
553-AAA1093

Developing an equipment layout plan for Chassis systems

Contents

This section contains information on the following topics:

Introduction

Take some time to plan the installation of the Chassis system. This preparation helps to make sure the system performs correctly. Develop a layout plan for the equipment to determine where you will position each system component.

Give consideration to the lengths of the different cables, so that you make the best use of available space. Refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for a description of Chassis system cable and wire specifications.

Preparation of the site according to the plan is very important. Make sure that the site is ready to accept the equipment. Make sure that items, such as power outlets and backboards, are installed correctly.

General layout guidelines

> ⚠️ **WARNING**
> Make sure that the mounting surface can support at least 100 lb (45 kg).

The following are the Meridian 1 PBX 11C Chassis installation options:

- wall installation
  - — vertically on a wall
  - — horizontally on a wall

- in a rack/cabinet

Each chassis measures 8.4 in. (213 mm) high by 17.2 in. (437 mm) wide by 12.8 in. (325 mm) deep.

Additional considerations for multiple-chassis systems

If you are combining Small Systems, you must adhere to the following minimum standards:

- A horizontal installation of Chassis system requires 10 inches of free space on either side of the chassis.

- A vertical installation of Chassis system requires 12 inches of free space on the card side and 6 inches of free space on the cable side of the chassis.

For multi-chassis systems, the following guidelines apply for both horizontal and vertical expansion:

- The maximum distance between the Main Chassis and each fiber Expansion Chassis is 1.8 mi (3 km).

- The minimum distance between the Main Chassis and the chassis expander, when mounted above one another (vertical expansion), is 4 in. (102 mm).

  Note: The equipment layout plans shown in this chapter are applicable to fiber-optic connected chassis installed within close proximity to each other (such as on the same wall). These layout guidelines are not as stringent if the chassis are located in separate rooms, on different floors, or in different buildings.

Installing horizontally or vertically on a wall

Figure 19 "Typical layout for installing the chassis on a wall in a horizontal position" (page 88) shows a typical wall layout, using BIX cross-connect equipment, for installing the chassis on a wall in a horizontal position. Figure 20 "Typical layout for installing the chassis on a wall in a vertical position" (page 89) shows a typical wall layout, using BIX cross-connect equipment, for installing the chassis on a wall in a vertical position. Use of other types of terminal blocks and equipment can change the layout. As a result, if required, adjust the height at which you place the chassis in relation to other equipment. If required, also adjust the distances between the power outlets and the backboard.

Use the following guidelines to position the system equipment on a wall:

- Nortel recommends that you fasten a 3/4 in. (20 mm) plywood (or other material like plywood) backboard to the surface of the wall. Fasten the Chassis system equipment to this backboard.

- When planning for a system with DTI/PRI capability, allow space on the backboard for the Channel Service Unit (CSU).

- Leave at least 6 in. (155 mm) above the mounting bracket and any obstruction (such as a pipe or conduit) so that there is room to lift the chassis on and off the bracket.

- Leave at least 12 in. (305 mm) between the top of a chassis and the ceiling to make sure that there is enough ventilation for the system.

- Leave 10 in. (255 mm) between the bottom of the lower chassis and the floor to prevent water damage.

- If you use the NT1R20 Off-Premise Station card, allow for correct installation (according to local practices).

- Make sure power outlets are within reach of each system chassis. See *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for cable and wire specifications.

If you are combining cabinets and chassis in a mix-and-match configuration, the following minimum standards must be followed:

- For horizontal installation, the chassis requires 10 inches of free space on either side of the chassis.

- For vertical installation, the chassis requires 12 inches of free space on the card side and 6 inches of free space on the cable side of the chassis.

Figure 19
Typical layout for installing the chassis on a wall in a horizontal position

Allow 24 in. (610 mm) at the top for miscellaneous equipment

Leave at least 2 in. (51 mm) between edge of backboard and corner of room

Draw 2 vertical lines 18.5 in. (470 mm) apart

Draw horizontal line for bottom hole of ear bracket

Distance from edge of backboard to isolated ground outlet 0 - 27 in. (686 mm)

Maximum of 21 in. (533 mm)
Minimum of 12 in. (305 mm)

Main Chassis

Chassis Expander

12 in. (305 mm)

10 in. (254 mm)

Each chassis measures 8.4 in. x 17.2 in. x 12.8 in. (213 mm x 437 mm x 325 mm)

Recommended lowest position for chassis installation is 10 in. (254 mm) from floor

553-9034

Note: Leave wall space for the cross-connect terminal.

Figure 20
Typical layout for installing the chassis on a wall in a vertical position

Leave at least 12 in. (305 mm) between edge of backboard and corner of room to allow for card insertion

Allow 24 in. (610 mm) at the top for miscellaneous equipment

Draw horizontal line for bottom hole of wall mount bracket

Main Chassis

Spacing is 21 in. (530 mm)

Chassis Expander

Distance from edge of backboard to isolated ground outlet 0 - 27 in. (686 mm)

12 in. (305 mm)

10 inches (254 mm)

Each chassis measures 8.4 in. x 17.2 in. x 12.8 in. (213 mm x 437 mm x 325 mm)

Recommended lowest position for chassis installation is 10 inches (254 mm) from floor

553-9035

Note: Leave wall space for the cross-connect terminal.

Installing in a rack/equipment cabinet

You can install the chassis and chassis expander in a 19-inch rack/equipment cabinet. There is also space in the rack/equipment cabinet for additional pieces of Data Communications Equipment. In a rack/equipment cabinet configuration, the lowest recommended installation position for the chassis is 1.75 inches (44 mm) from the floor. See .

> *Note:* The 19-inch rack/equipment cabinet does not come with the Chassis system.

Figure 21
Typical layout for installing the chassis in a rack/equipment cabinet
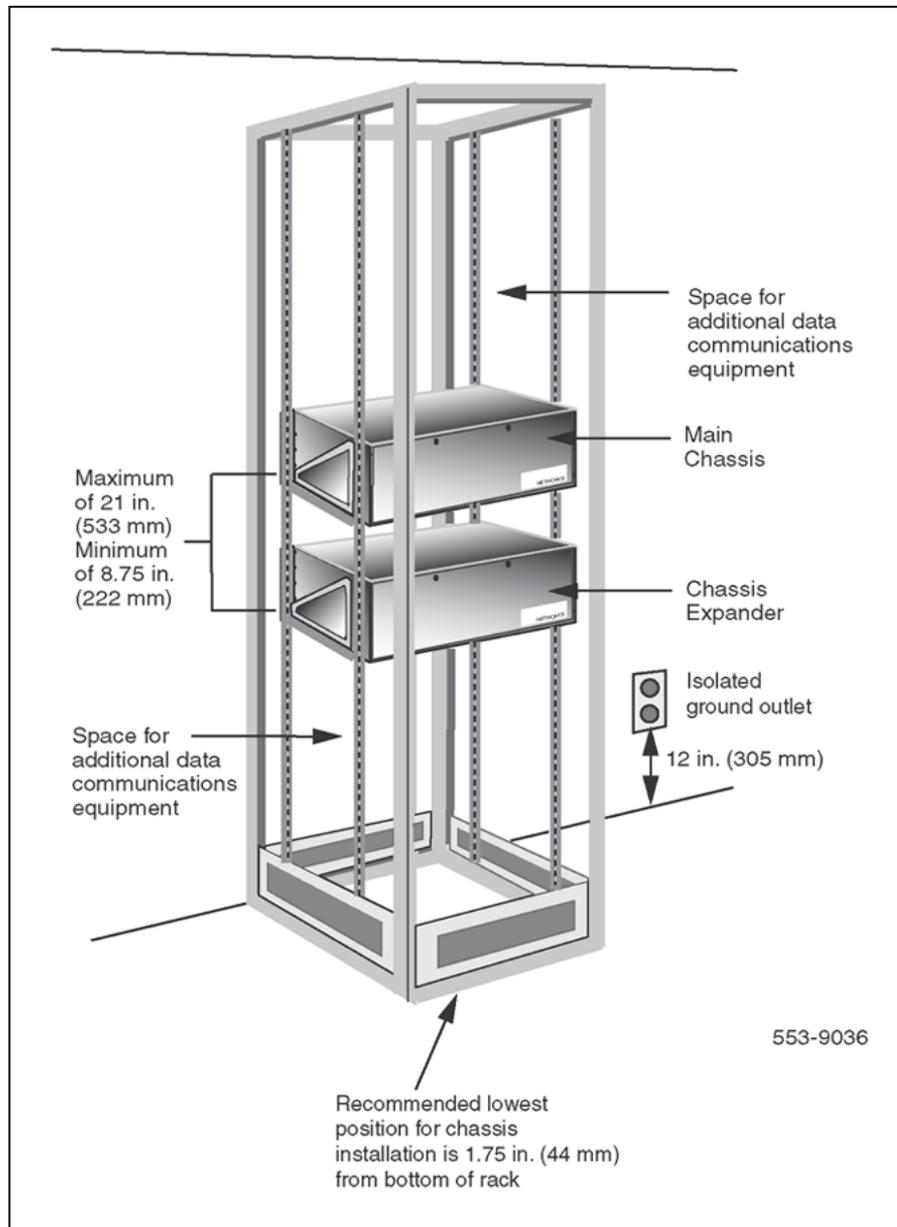
> *Note:* Leave wall space for the cross-connect terminal.

System controller cards and software daughterboards

Contents

This section contains information on the following topics:

Introduction

This chapter describes the components and features of the NTDK20 Small System Controller (SSC) card, including its software daughterboards.

NTDK20 Small System Controller card

The NTDK20 Small System Controller (SSC) card controls call processing, stores system and customer data, and provides various expansion interfaces (see Figure 22 "NTDK20 SSC card" (page 92)). The SSC card includes the following components and features:

- software daughterboard memory, DRAM, and backup memory

- two expansion daughterboard interfaces

- one PC Card socket

- three Serial Data Interface (SDI) network interfaces

- 32 channels of Conferencing (64 if two single-port expansion daughterboards are present, or 96 if two dual-port expansion daughterboards are present)

- one Ethernet (10 Mbps interface) port

- 30 channels of tone and digit switch (TDS) and a combination of 8 Digitone receivers (DTR) or dial-tone detectors (XTD)

- Networking and Peripheral Signaling

- additional tone service ports (four units of MFC/MFE/MFK5/MFK6/MFR or eight DTR/XTD units)

Figure 22 "NTDK20 SSC card" (page 92) shows the SSC card and its components.

Figure 22
NTDK20 SSC card
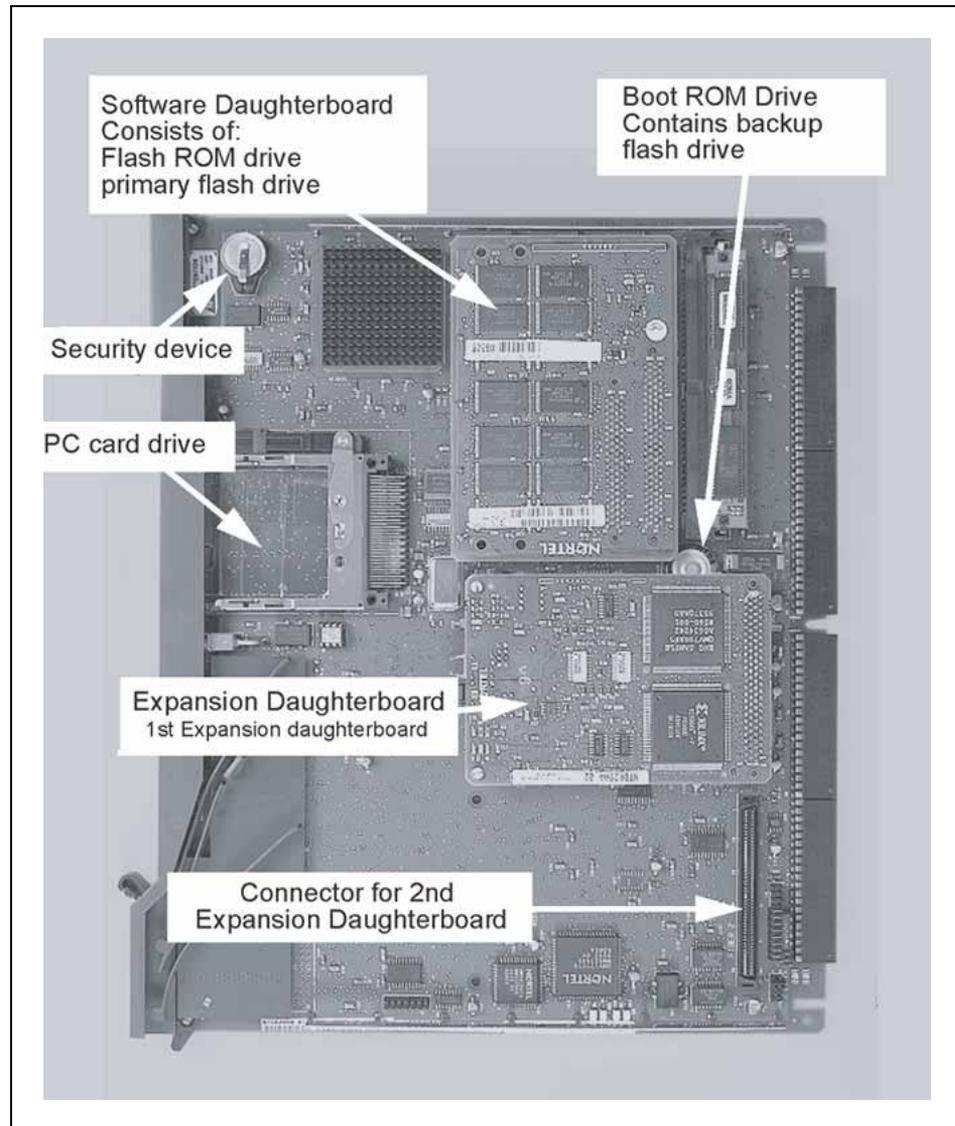
PC Card interface

The NTDK20 SSC card has a PC Card interface through a socket located on its faceplate. The PC Card socket can accommodate a Software Delivery card used for software upgrading and as backup media.

SDI network interfaces

The NTDK20 SSC card contains three SDI network interfaces used to connect on-site terminals or remote terminals through a modem. The default settings on the network interfaces are as shown in Table 20 "Default SDI network interface settings on the NTDK20 SSC card" (page 93).

Table 20
Default SDI network interface settings on the NTDK20 SSC card

TTY Port	Baud rate	Data bits	Stop bits	Parity	Use
0	Set by a DIP switch (default 19 200)	8	1	None	MTC/SCH/BUG
1	1200	8	1	None	MTC/SCH/BUG
2	1200	8	1	None	MTC/SCH/BUG

Conferencing

Thirty-two conference channels are provided by the NTDK20 SSC card's conference devices. Conference capability can be increased by mounting expansion daughterboards on the NTDK20 SSC card. Each daughterboard increases the total number of conference channels by 16: the maximum number of conference ports is 64.

Each conference device provides 16 ports of conferencing capabilities (one conference participant for each port). A conference call can have three to six participants. To illustrate, you can have a maximum of five 3-party conferences for each device, or two 6-party conferences plus one 3-party conference. You cannot conference between conference devices.

Tone services

The NTDK20 SSC card incorporates the functions of the existing NTAK03 TDS/DTR, NT5K20 XTD, and NT5K48 XTD cards.

IP expansion 10BaseT port

The Small System provides one 10 Mbps Ethernet connection to a Local Area Network (LAN). The 10BaseT Ethernet network interface available on the SSC of an IP Expansion Cabinet or Chassis is functional. However, the Ethernet network interface on the IP Expansion Cabinet or Chassis does not have a default IP configuration. This means that you must perform the IP port configuration before it can be used.

Nortel does not recommend using the remote 10BaseT port in normal mode, as maintenance or alarm management are not available. In survival mode it assumes the system-level configuration of the Main Cabinet or Chassis port.

External connection to the Ethernet network interface is provided by a 50-pin connector located in the Main Cabinet. An NTDK27 Ethernet Adaptor cable adapts this 50-pin connector to the standard 15-pin AUI interface for a Medium Access Unit (MAU).

The Chassis system has a standard 15-pin AUI interface for a MAU to be connected.

Software daughterboards

The SSC card's software daughterboard performs a significant portion of system software storage and data processing for the Small System.

Memory

The majority of system and customer-configured data is both controlled and stored on the flash ROM of the SSC card, which comprises the program store. An active and a backup copy of customer data is also kept on the flash ROM.

Additional memory, referred to as Dynamic Random Access Memory (DRAM), stores and processes temporary automated routines and user-programmed commands. The SSC card also retains a copy of customer files in the event of data loss, in an area called the backup flash drive.

NTTK25 Software daughterboard

The NTTK25 is a 48 Mbyte daughterboard comprised of program store and primary flash.

- The program store holds 32 Mbyte of ROM memory, comprising operating system data and overlay programs.

- The primary flash resides on the remaining 16 Mbyte of flash space, which provides 14.7 Mbyte of formatted storage. This is used to store customer data, patches, log files and other system data.

Boot code

The boot code on existing SSC cards must be NTDK34FA Release 7 or later to support the NTTK13 or NTTK25 Software daughterboards. Nortel advises that the SSC boot code must be confirmed or upgraded to the latest version every time the software is installed or upgraded. The boot code can be found on the programmed PC Card.

Note: New Small Systems have the latest version of software preprogrammed on the software daughterboard.

System data

Other system data, such as the Secure Storage Area (SSA), also resides in the flash. The SSA holds data that must survive power-downs.

BootROM is a 2 Mbyte storage device located on the SSC card's motherboard. It is comprised of boot code, system data, patch data, and the backup copy of the primary flash drive's customer database.

DRAM

The NTDK20HA SSC card is equipped with 32 Mbyte of temporary memory space called DRAM. DRAM functions much like RAM on a computer system, whereby system and user files are stored while the system is up and running. DRAM on the Small System stores operating system files, overlay data, patch codes, and the active copy of the customer database.

Memory requirements for CS 1000 Release 4.5

For CS 1000 Release 4.5 software, the minimum requirements are:

* 32 Mbyte DRAM

* 16 Mbyte primary flash

* 32 Mbyte program store

    Note: The NTDK97 Mini System Controller (MSC) card, which has 16 Mbyte DRAM, will support certain limited Chassis system configurations. Refer to Note 3 at the bottom of Table 21 "System controller card requirements for Small Systems" (page 96) for details.

An NTDK20 SSC card with 32 Mbyte DRAM and equipped with an NTTK13 or NTTK25 Software daughterboard meets the minimum requirements.

New systems are shipped with the NTDK20HA SSC card equipped with the NTTK25BA Software daughterboard.

Existing Cabinet or Chassis systems equipped with earlier vintages of the SSC card must:

* upgrade the SSC card using the NTDK19 SSC upgrade kit

* if necessary, upgrade to the NTTK25 Software daughterboard

    Note: If you are installing a new software daughterboard on an older vintage SSC card, you must upgrade the boot code on the SSC card before installing the new daughterboard.

In existing Chassis systems equipped with the MSC card, replace the MSC card with a suitable SSC card, to eliminate any potential issues.

Table 21 "System controller card requirements for Small Systems" (page 96) summarizes the system controller card requirements for existing Small System configurations upgrading to CS 1000 Release 4.5.

Table 21
System controller card requirements for Small Systems

System configuration	Main Cabinet/Chassis	Expansion Cabinet/Chassis
Single cabinet	• NTDK20HA SSC • Upgraded NTDK20AB-GA SSC (see Note 2)	n/a
Fiber expansion Cabinet system	• NTDK20HA SSC • Upgraded NTDK20AB-GA SSC (see Note 2)	n/a (use Fiber Receiver card)
IP expansion Cabinet system	• NTDK20HA SSC • Upgraded NTDK20EA-GA SSC	NTDK20HA SSC Upgraded NTDK20AB-GA SSC (see Note 2)
Single chassis	• NTDK20HA SSC • Upgraded NTDK20EA-GA SSC • NTDK97AD MSC (see Note 3)	n/a
Fiber expansion Chassis system	• NTDK20HA SSC • Upgraded NTDK20EA-GA SSC	n/a (use Fiber Receiver card)
IP expansion Chassis system	• NTDK20HA SSC • Upgraded NTDK20EA-GA SSC	• NTDK20HA SSC • Upgraded NTDK20EA-GA SSC

Note 1: The term *upgraded NTDK20xx* refers to an SSC card vintage NTDK20xx on which the NTDK19 Upgrade Kit is installed.

Note 2: NTDK20AB/CA Rls. 1-5 is not supported on upgrades. Refer to Product Bulletin P-2004-0067-Global.

Note 3: With CS 1000 Release 4.5 software, the maximum capacity of the MSC in a pure IP solution is 1000 IP TNs and 2000 Corporate Directory entries. Maximum capacity in a TDM solution is 160 TDM TNs and 2000 Corporate Directory entries. Maximum capacity in a mixed TDM/IP solution is 144 TDM TNs, 300 IP TNs, and 2000 Corporate Directory entries.

Security device

A security device is required on the NTDK20 SSC card of the Main and
all IP Expansions. For more information, see "Security device for the IP
Expansion" (page 102).

System expansion

Contents

This section contains information on the following topics:

Introduction

Both the Cabinet system and the Chassis system support IP and fiber-optic expansion. Up to four Expansion Cabinets or Chassis can be connected to the Main with 100BaseT/F cable or fiber-optic cable and located up to 3 km (1.8 mi) from the Main. With the introduction of IP connectivity, IP expansion can be distributed over a campus data network.

Note 1: The distance between cabinets or chassis is determined by the length of the fiber-optic cable.

Note 2: With 100baseF expansion daughterboards and third-party converters, the distance can be extended to more than 20 km.

This section describes the IP and fiber expansion daughterboards, fiber receiver cards, cabling, and other related equipment required for IP or fiber expansion.

Expansion daughterboards

Expansion daughterboards mounted on the NTDK20 SSC card (see Figure 22 "NTDK20 SSC card" (page 92)) allow the connection of the Main Cabinet or Chassis to Expansion Cabinets or Chassis in multi-cabinet/chassis Small Systems. Each port on each daughterboard also provides an additional 16-channel conference loop and up to three SDI network interfaces on the Expansion Cabinet or Chassis.

Table 22 "Expansion daughterboards" (page 100) summarizes the network interfaces, cables, and connection data for Fiber and IP Expansion daughterboards.

Table 22
Expansion daughterboards

Daughterboard	Number of network interfaces	Cable type	Max. distance between Main and Expansion Cabinets/Chassis
NTDK22	one	A0632902 fiber-optic plastic cable	10 m (33 ft)
NTDK84	two		
NTDK24	one	glass fiber-optic cable	3 km (1.8 mi)
NTDK85	two		
NTDK79	one	single-mode glass fiber-optic cable	
NTDK99	one	100baseT cable (see "EMC grounding clip" (page 103))	100 m (328 ft), or over 20 km (12 mi) with a third-party converter
NTDK83	two		
NTTK01	one	100baseF fiber-optic cable	2 km (1.2 mi), or over 20 km (12 mi) with a third-party converter
NTTK02	two		

IP Expansion daughterboards

For IP expansion, you must install the IP Expansion daughterboard in Connector #2 (the lower connector) on the SSC card to ensure clock synchronization.

There are four types of IP Expansion daughterboards:

- NTDK99 single-port 100BaseT IP Expansion daughterboard — provides connectivity to one IP Expansion Cabinet or Chassis located within 100 m (328 ft)

- NTDK83 dual-port 100BaseT IP Expansion daughterboard — provides connectivity to two IP Expansion Cabinets or Chassis located within 100 m (328 ft)

- NTTK01 single-port 100BaseF IP Expansion daughterboard — provides connectivity to one IP Expansion Cabinet or Chassis located within 2 km (1.2 mi), using glass fiber-optic cable

- NTTK02 dual-port 100BaseF IP Expansion daughterboard — provides connectivity to two IP Expansion Cabinets or Chassis located within 2 km (1.2 mi), using glass multimode optic cable

  Note: Third-party media conversion devices can be used to extend the range of IP Expansion Cabinets or Chassis from the Main Cabinet or Chassis.

Figure 23 "Expansion daughterboards" (page 102) shows examples of the IP Expansion daughterboards.

Figure 23
Expansion daughterboards
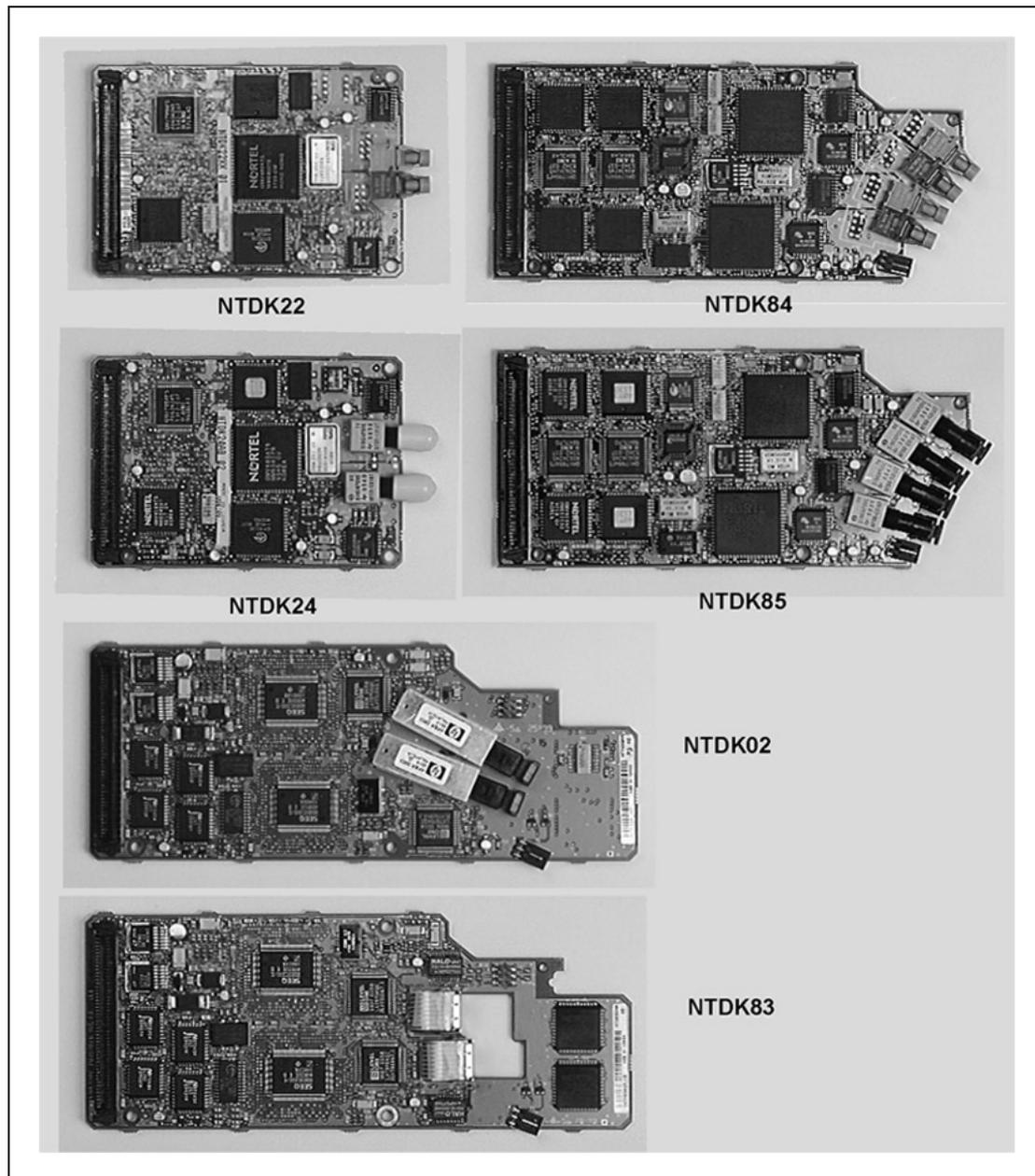
Security device for the IP Expansion

The SSC card on the Small System must contain an NTDK57AA security device, which is keycoded to match the NTDK57DA security device on each IP Expansion.

This maintains the requirement of a single keycode for each Small System with survivable IP Expansion Cabinets or Chassis. The main objectives of this security scheme are the following:

Step	Action
1	Allow the system to operate as a single system when all links are up.
2	Allow the survivable IP Expansion Cabinet or Chassis to continue operating with its existing configuration in the event of a failure of the Main Cabinet or Chassis, or of the link to the Main.
3	Prevent users from configuring or using more TNs or features than have been authorized.

—End—

The IP Expansion Cabinet or Chassis security device provides the following capabilities at the Expansion Cabinet or Chassis:

- System software can be installed but no calls will be processed or features activated until communication with a Main has been established and a match between the security ID of the Main and the IP Expansion Cabinet or Chassis has been confirmed.

- System software can be upgraded.

- Local datadump is not permitted, as well as all LD 43 and LD 143 commands.

EMC grounding clip

Cabinets and chassis connected with 100BaseT IP connectivity must route the cables though the EMC grounding clip. This ensures electrical contact between the ground rail and 100BaseT cable for EMC containment.

The NTDK41AA EMC grounding clip is used on the Cabinet system on each IP Expansion Cabinet. Figure 24 "EMC grounding clip on Main Cabinet" (page 104) shows the EMC grounding clip on the Main cabinet.

Figure 24
EMC grounding clip on Main Cabinet
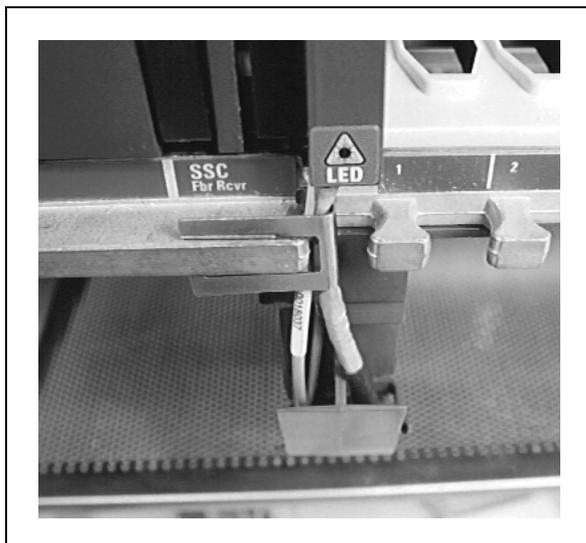
The NTTK43AA EMC grounding clip is used on the Chassis and IP
Expansion Chassis (see
).

Figure 25
EMC grounding clip on the Chassis
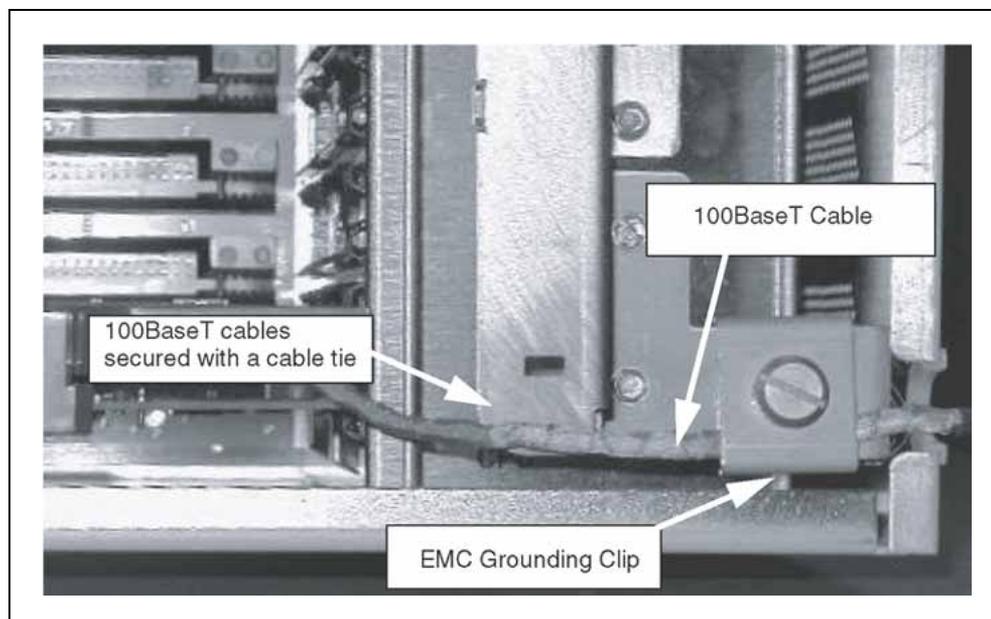
> ⚠ **WARNING**
> Use of the EMC grounding clip is required for EMC compliance.

For further information or installation instructions, refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.

Media conversion devices

Third-party media conversion devices can be used to extend the range of the 100BaseT and 100BaseF IP solutions. However, use caution when extending the length of cable used in the point-to-point configuration. The Round Trip Delay (RTD) parameters specified in *Converging the Data Network with VoIP Fundamentals (NN43001-260)* must not be exceeded.

Fiber Expansion daughterboards

Fiber Expansion daughterboards mounted on the NTDK20 SSC card allow the connection of fiber-optic cables from the Main Cabinet or Chassis to Expansion Cabinets or Chassis in multi-cabinet systems. Each daughterboard also provides an additional 16-channel conference loop and one SDI port at the Expansion Cabinet.

There are five types of Fiber Expansion daughterboard:

- "NTDK22 Fiber Expansion daughterboard (single-port)" (page 105)
- "NTDK84 Fiber Expansion daughterboard (dual-port)" (page 105)
- "NTDK24 Fiber Expansion daughterboard (single-port)" (page 106)
- "NTDK85 Fiber Expansion daughterboard (dual-port)" (page 106)
- "NTDK79 Fiber Expansion daughterboard (single-port)" (page 106)

NTDK22 Fiber Expansion daughterboard (single-port)

The NTDK22 single-port Fiber Expansion daughterboard is used when the Expansion Cabinet or Chassis is within 10 m (33 ft) of the Main Cabinet or Chassis. It connects to one A0632902 plastic multimode fiber-optic cable. One NTDK22 daughterboard is required to support each fiber Expansion Cabinet or Chassis located within 10 m (33 ft) of the Main Cabinet or Chassis.

NTDK84 Fiber Expansion daughterboard (dual-port)

The NTDK84 dual-port Fiber Expansion daughterboard has the same features as the NTDK22 except that it can interface with two Expansion Cabinets or Chassis. It connects to two A0632902 plastic multimode

fiber-optic cables. One NTDK84 daughterboard is required to support two fiber Expansion Cabinets or Chassis located within 10 m (33 ft) of the Main Cabinet or Chassis.

NTDK24 Fiber Expansion daughterboard (single-port)
The NTDK24 single-port Fiber Expansion daughterboard is used when the Expansion Cabinet or Chassis is up to 3 km (1.8 mi) of the Main Cabinet or Chassis. It connects to one glass multimode fiber-optic cable, which is dedicated to the Small System. One NTDK24 daughterboard is required for to support each Expansion Cabinet or Chassis located up to 3 km (1.8 mi) of the Main Cabinet or Chassis.

NTDK85 Fiber Expansion daughterboard (dual-port)
The NTDK85 dual-port Fiber Expansion daughterboard has the same features as the NTDK24 except that it can interface with two Expansion Cabinets or Chassis. It connects to two glass multimode fiber-optic cables. One NTDK85 daughterboard is required to support two Expansion Cabinets or Chassis located up to 3 km (1.8 mi) of the Main Cabinet or Chassis.

NTDK79 Fiber Expansion daughterboard (single-port)
The NTDK79 single-port Fiber Expansion daughterboard has the same features as the NTDK24 except that it connects to single-mode glass fiber-optic cable. One NTDK79 daughterboard is required to support each Expansion Cabinet or Chassis located up to 3 km (1.8 mi) of the Main Cabinet or Chassis.

> *Note:* When both Fiber and IP Expansion daughterboards coexist in a system, configure the Fiber Expansion daughterboard as expansions 1 and 3 and the IP Expansion daughterboard as expansions 2 and 4.

Fiber Receiver cards
Fiber Receiver cards installed in the Fbr Rx slot (slot 0) of Expansion Cabinets allow the connection of fiber-optic cables between the Main Cabinet or Chassis and up to four fiber Expansion Cabinets/Chassis.Figure 26 "Fiber Receiver card in fiber Expansion Cabinet (NTDK23 shown)" (page 108) shows a Fiber Receiver card in a fiber Expansion Cabinet.

There are three versions of the Fiber Receiver card, each of which has a corresponding fiber daughterboard:

- "NTDK23 Fiber Receiver card" (page 107)
- "NTDK80 Fiber Receiver card" (page 107)
- "NTDK80 Fiber Receiver card" (page 107)

NTDK23 Fiber Receiver card

The NTDK23 Fiber Receiver card is used when the Expansion Cabinet is within 10 m (33 ft) of the Main Cabinet. It connects to one A0618443 fiber-optic cable. One of these cards is required for each Expansion Cabinet located within 10 m (33 ft) of the Main Cabinet.

The NTDK23 Fiber Receiver card works in conjunction with either an NTDK22 or an NTDK84 Fiber Expansion daughterboard in the Main Cabinet.

NTDK25 Fiber Receiver card

The NTDK25 Fiber Receiver card is used when the Expansion Cabinet is located up to 3 km (1.8 mi) of the Main Cabinet. It connects to one multimode glass fiber-optic cable, which is dedicated to the system. One of these cards is required for each Expansion Cabinet located up to 3 km (1.8 mi) of the Main Cabinet, connected by multimode fiber-optic cable.

The NTDK25 Fiber Receiver card works in conjunction with either an NTDK24 or an NTDK85 Fiber Expansion daughterboard in the Main Cabinet.

Note: The NTDK24 supports only multimode glass fiber-optic cable.

NTDK80 Fiber Receiver card

The NTDK80 Fiber Receiver card is used when the Expansion Cabinet is located more than 3 km (1.8 mi) from the Main Cabinet. It connects to one single-mode glass fiber-optic cable. One of these cards is required for each Expansion Cabinet located more than 3 km (1.8 mi) from the Main Cabinet.

The NTDK80 Fiber Receiver card works in conjunction with an NTDK79 Fiber Expansion daughterboard in the Main Cabinet.

Note: The NTDK80 supports only single-mode glass fiber-optic cable.

SDI port

Each Fiber Receiver card supports one Serial Data Interface (SDI) port allowing remote TTY access.

Figure 26
Fiber Receiver card in fiber Expansion Cabinet (NTDK23 shown)
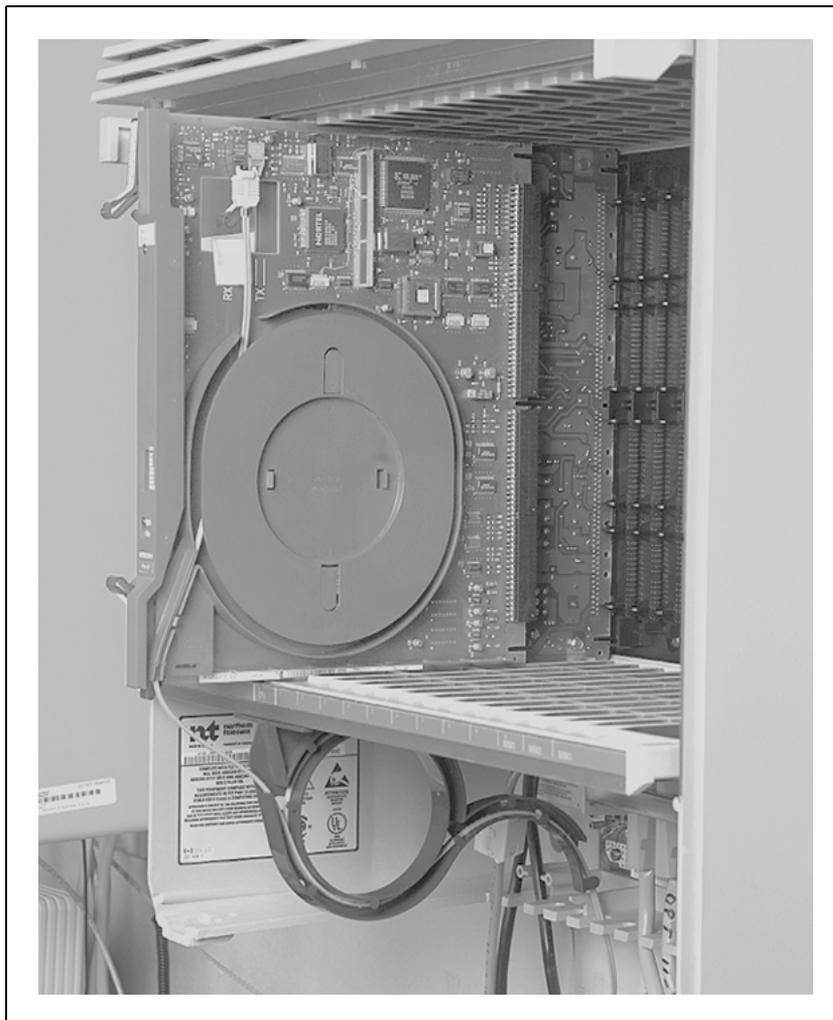
Expansion cabling

Note 1: Any reference to cabinets in this section equally applies to chassis if you are using them in your fiber or IP expansion system.

Note 2: The MFI and EFI units used with Option 11E to interface with fiber-optic cable cannot be used with the Small System.

IP connector cables

IP-expanded systems require the cables listed in Table 23 "IP connector cables" (page 109).

Table 23
IP connector cables

Daughterboards	Cable	Cable description
NTDK83 and NTDK99 100baseT IP	NTTK34AA/ AO793725	10m RJ–45 CAT 5 Ethernet cable
	NTDK8305 / AO781621	2m STP CAT 5 extension cable
NTTK01 and NTTK02 100baseF IP	AO817052	5 m MT-RJ to ST cable.
	A0346816	ST fiber coupler
	AO817055	10m MT-RJ to MT-RJ fiber extension cable

Fiber-optic cable

Cabinets can be located up to 3 km (1.8 mi) from the Main Cabinet by using fiber-optic cable. There are three types of connections:

- plastic fiber-optic cable

- glass fiber-optic cable

- "IP connector cables" (page 109)

Plastic fiber-optic cable (multimode)

The A0632902 fiber-optic cable is a 10 m (33 ft) plastic fiber cable that is used when the Expansion Cabinet is located 10 m (33 ft) or less from the Main Cabinet. This cable comes equipped with a connector on each end, which connect to either the NTDK22 or the NTDK84 Fiber Expansion daughterboard in the Main Cabinet and to the NTDK23 Fiber Receiver card in the Expansion Cabinet. This cable, which is the only cable that can be used for this purpose, is not intended to be cut or altered in the field. Cable management devices store excess cable in the cabinets.

Glass fiber-optic cable

Glass fiber-optic cable (multimode or single-mode, depending on interface cards) is required when the cable length between the Main Cabinet and an Expansion Cabinet is up to 3 km (1.8 mi).

Note: The distance between the Main Cabinet and Expansion Cabinet is determined by the length of the fiber-optic cable — not by linear distance.

This glass fiber cable, which is supplied by a local telephone company or other facilities provider, must be dedicated to the system and cannot be shared with other services.

A connector is required on each end of the cable to connect to the NTDK24 (multimode), NTDK85 (multimode), or NTDK79 (single-mode) Fiber Expansion daughterboard in the Main Cabinet and to the NTDK25 (multimode) or NTDK80 (single-mode) Fiber Receiver card in the Expansion Cabinet. Cable management devices store excess cable in the cabinets.

> *Note:* The Small System fiber-optic link for distances up to 3 km (1.8 mi) uses the industry standard 62.5/125 µm glass multimode duplex cable or 8/125 µm glass single-mode cable with ST-type connectors.

The type of cable used depends on the type of installation and any local building codes.

Table 24 "Multimode glass optical cable requirements" (page 110) Table 25 "Single-mode glass optical cable requirements" (page 111) list the minimum optical requirements for multimode and single-mode glass fiber-optic cable.

Table 24
Multimode glass optical cable requirements

Parameter	Minimum	Typical	Maximum	Units
Glass fiber cable length			3.0 (see Note)	km
Cable attenuation @1300 nm		1.5	2.0	dB/km
Modal bandwidth @1300 nm	200	500		MHz * km
Chromatic dispersion @1300 nm		6		ps/nm * km
Typical 3 dB optical bandwidth		180		MHz * km
Splice attenuation		0.1	0.2	dB

Note: The fiber link should be limited to a maximum length of 3 km, even though with many optical cables the optical power budget could support greater lengths. To guarantee reliable operation, a bandwidth of 150% should be maintained. If the link is increased beyond the 3 km length, the 150% margin deteriorates, resulting in a possible link malfunction.

Parameter	Minimum	Typical	Maximum	Units
Coupler attenuation		0.5	0.5	dB
Power budget			8	dB

Note: The fiber link should be limited to a maximum length of 3 km, even though with many optical cables the optical power budget could support greater lengths. To guarantee reliable operation, a bandwidth of 150% should be maintained. If the link is increased beyond the 3 km length, the 150% margin deteriorates, resulting in a possible link malfunction.

Table 25
Single-mode glass optical cable requirements

Parameter	Minimum	Typical	Maximum	Units
Cable attenuation		0.4	0.5	dB/km
Splice attenuation		0.1	0.2	dB
Coupler attenuation		0.2	0.5	dB
Power budget			5	dB
Glass fiber length			3 (see Note)	km

Note: The fiber link should be limited to a maximum length of 3 km, even though with many optical cables the optical power budget could support greater lengths. To guarantee reliable operation, a bandwidth of 150% should be maintained. If the link is increased beyond the 3 km length, the 150% margin deteriorates resulting in a possible link malfunction.

Environment

The fiber Expansion daughterboards and Fiber Receiver cards are subject to the environmental conditions shown in Table 26 "Environmental conditions" (page 111).

Table 26
Environmental conditions

	Operating	Storage
Ambient temperature	0° C to 50° C (32° F to 122° F)	-45° C to 70 ° C (-49° F to 158° F)
Relative humidity	5% to 95%	0% to 95%

Small System distribution over a data network

Contents

This section contains information on the following topics:

Introduction

Small System IP expansion allows connectivity of IP Expansion Cabinets or Chassis either point-to-point or over a distributed campus data network. The campus data network connectivity is provided through IP daughterboards in the Main and IP Expansion Cabinets or Chassis.

Note: This section is not applicable for Signaling Server or Voice Gateway Media Cards.

Figure 27 "IP expansion configuration of cabinets or chassis over a campus data network" (page 114) provides an example of Main and IP Expansion Cabinets and Chassis connected over a campus data network using both 100BaseT and 100BaseF connectivity.

In order to satisfy PBX voice quality requirements, engineering guidelines are imposed on the campus data network. Refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

Note: Contact your local Data Administrator to obtain specific IP information.

Figure 27
IP expansion configuration of cabinets or chassis over a campus data network

553-120A
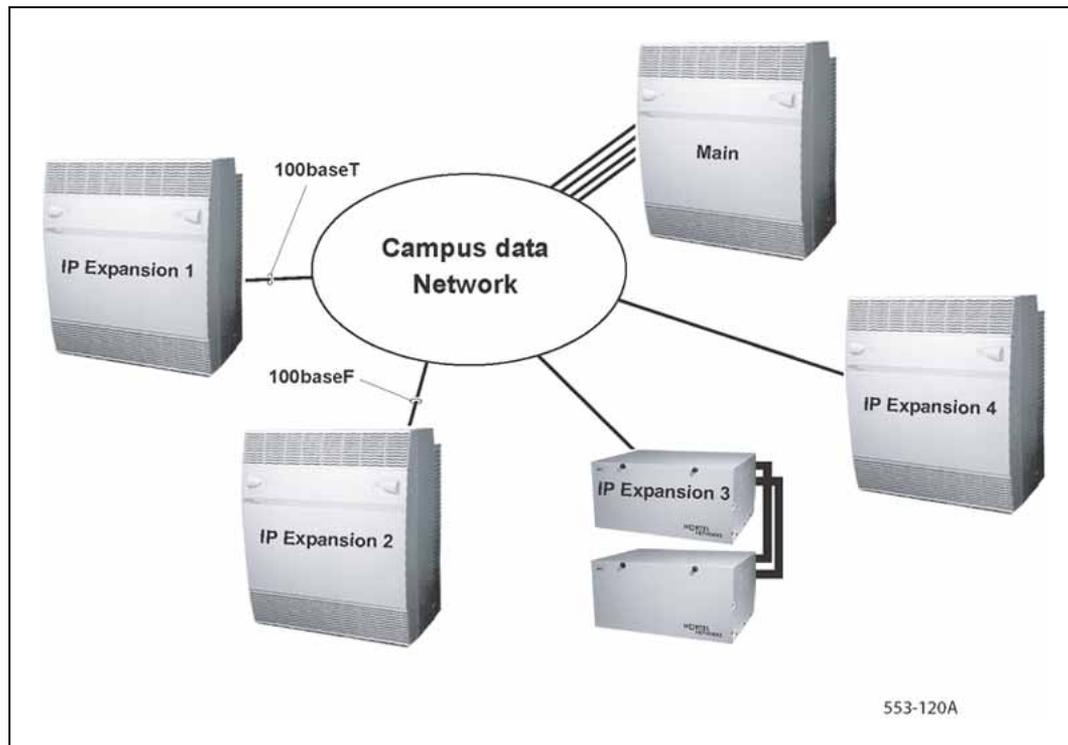
Monitoring IP link voice Quality of Service for IP Expansion Cabinets or Chassis

Behavioral characteristics of the network are dependent on factors such as Round Trip Delay (RTD), queuing delay in the intermediate nodes, packet loss, and available bandwidth. The service level of each IP link is measured and maintained on the Main Cabinet or Chassis for IP expansion operation. Information for latency and packet loss is collected from the hardware and processed.

Based on system configured thresholds, the level of service is derived and reported to the craftsperson with the `PRT QOS <cab#>` command in LD 117. See *Software Input Output Administration (NN43001-611)* and *Software Input Output Reference - Maintenance (NN43001-711).*

Refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for more information on the Data Network Ratings (Excellent, Good, Fair, Poor) and their parameter values for network delay.

Circuit cards

Contents

This section contains information on the following topics:

Introduction

This chapter describes various optional circuit cards commonly used in Small Systems. This chapter does not include information on the NTDK20 Small System Controller (SSC) card or the Fiber Receiver cards. For information on these cards, refer to "NTDK20 Small System Controller card" (page 91) and "Fiber Receiver cards" (page 106).

Refer to *Circuit Card Reference (NN43001-311)* for full descriptions of country-specific IPE cards.

Optional circuit cards

NTAK02 SDI/DCH card

The NTAK02 SDI/DCH switches allow you to configure the four SDI network interfaces on the Main Cabinet or Chassis as a combination of the following:

- SDI
- ESDI, or
- DCH/DPNSS

The NTAK02 SDI/DCH card uses jumper plugs to configure the RS232/RS422 interfaces as one of the following:

- DTE
- DCE

Note 1: In a Cabinet system, this circuit card can be installed in both the Main Cabinet and IP Expansion Cabinets. In a Chassis system, install this circuit card in the Chassis only.

Note 2: Only DCH is supported in IP Expansion cabinets. ESDI, AML, and TTY are not supported.

NT8D14 Universal Trunk card

The Universal Trunk card provides eight analog trunks.

NT8D15 E and M Trunk card

The E&M trunk card provides four trunks which can function as 2W E&M, 4W E&M, and Paging.

NTAG26 XMFR card

The NTDK20 SSC card provides the functionality of the Extended Multi-Frequency Receiver (XMFR) card. However, this card can exist with the SSC card if you want to access additional XMFR capability.

The XMFR card receives multifrequency (MF) digit information. Connections are made between a PBX and a Central Office (CO). Through the Intelligent Peripheral Equipment (IPE) MF Receiver, the Small System supports features such as Automatic Number Identification (ANI), Meridian 911 (M911), and Feature Group D (FGD).

NT5K21 XMFC card

The NTDK20 SSC card provides the functionality of the Extended Multi-Frequency Compelled Sender/Receiver (XMFC) card. However, the XMFC card can coexist with the SSC card to access extra XMFC capability.

The XMFC card provides four channels of R2 Standard signaling capability.

NT1R20 Off-Premise Station analog line card

The Off-Premise Station (OPS) analog line card provides eight OPS lines.

NTDK16 48-port Digital Line Card

The NTDK16 48-port Digital Line Card provides an interface to a maximum of 48 digital integrated voice and data sets in a Chassis system. The NTDK16 Digital Line Card is functionally equivalent to three NT8D02 Digital Line Cards.

Note 1: You can only place the NTDK16 Digital Line Card in slot 4 of the chassis.

Note 2: The Chassis system does not require the NTDK16 Digital Line Card to operate.

NT8D02 Digital Line Card

The NTDK16 48-port Digital Line Card is functionally equivalent to three NT8DO2 Digital Line Cards. However, the Chassis system also supports the NT8D02 Digital Line Card.

Digital Trunk cards

Small Systems supports the following digital trunk cards:

- NTAK10 2.0 Mbit DTI

- NTAK79 2.0 Mbit PRI

- NTBK22 MISP

- NTBK50 2.0 Mbit PRI

- NTRB21 1.5 Mbit DTI/PRI

- NT6D70 SILC (when used as a clock controller)

Note: In a Chassis system, the digital trunk cards can be installed only in slots 1–3 of the chassis.

Voice Gateway Media Card

The Voice Gateway Media Card (VGMC) supports IP Phone types i2001, i2002, i2004 and i2050. The following IP Phones are supported:

- Nortel IP Phone 2001

- Nortel IP Phone 2002

- Nortel IP Phone 2004

- Nortel IP Phone 2007

- Nortel IP Audio Conference Phone 2033

- Nortel IP Softphone 2050

- Nortel Mobile Voice Client (MVC) 2050

- WLAN Handsets 2210/2211

The VGMC provides a communication gateway between the IP data network and the Small System. The IP Phone translates voice into data packets for transport using Internet Protocol (IP). The IP Phone uses the customer's IP network to communicate with the VGMC and the optional Dynamic Host Configuration Protocol (DHCP) server. A DHCP server is used to provide the required information needed to enable the IP Phone network connection and to connect it to the VGMC.

There are three types of VGMCs:

- ITG-P 24-port line card (occupies 2 slots)

- Media Card 8-port line card (occupies 1 slot)

- Media Card 32-port line card (occupies 1 slot)

If a Media Card 32-port card, a Media Card 8-port card, or an ITG-P 24-port card is running IP Line 4.5 software, it is known as a Voice Gateway Media Card.

The VGMC plugs into the Small System's Main or IP Expansion Cabinet or Chassis. The VGMC communicates with the SSC card over the interface using 10BaseT connectivity.

The VGMC is administered using multiple management interfaces including:

- IP Line 4.5, a Graphical User Interface (GUI) provided by Optivity Telephony Manager (OTM) 2.2

- Command Line Interface (CLI)

- administration and maintenance overlays

- web browser interface provided by Element Manager

Refer to *IP Line Fundamentals (NN43001-500)* for further information about Voice Gateway Media Cards and the IP Line application.

SDI network interfaces

Contents

This section contains information on the following topics:

Introduction

This chapter describes the network interfaces on the Small System. Serial Data Interface (SDI) network interfaces are used to connect devices, such as terminals and modems to the Small System. The two types of SDI network interfaces supported are:

- Data Terminal Equipment (DTE); typically a TTY or computer
- Data Communication Equipment (DCE); typically a modem

SDI network interfaces are found on the SSC card, the optional TDS/DTR card, and the optional SDI/DCH card. An additional SDI network interface is located on the Fiber Receiver card to allow remote TTY access.

The possible Small System SDI network interface configurations are summarized in Table 27 "SDI network interface configurations" (page 119).

Table 27
SDI network interface configurations

Circuit Card	Number of ports	DTE	DCE	RS232	RS422
SSC NTDK20	3	Yes	No	Port 0	No
SDI/DCH	4	Ports	Ports	Ports	Ports 1/3

Circuit Card	Number of ports	DTE	DCE	RS232	RS422
NTAK02		0/1/2 /3	0/1/2 /3	0/1/2 /3	
NTDK23 Fbr Rcvr card	1	Yes	No	Yes	No
NTDK25 and NTDK80 Fbr Rcvr card	1	Yes	No	Yes	No

System controller cards

The NTDK20 Small System Controller card (used with Cabinet and Chassis systems) is equipped with three SDI network interfaces.

Each network interface can be used to connect a modem or terminal to the system. If connection to a terminal is desired, an A0378652 NO modem (NULL modem without hardware handshaking) is required.

For the Cabinet system, the SDI network interface is located at the bottom rear of the cabinet next to the connectors to the cross-connect terminal. (An NTBK48 three-port cable is required to connect to system equipment.) Refer to Figure 28 "Cabinet SDI cable connector" (page 120).

Figure 28
Cabinet SDI cable connector

NTBK48 cable connection
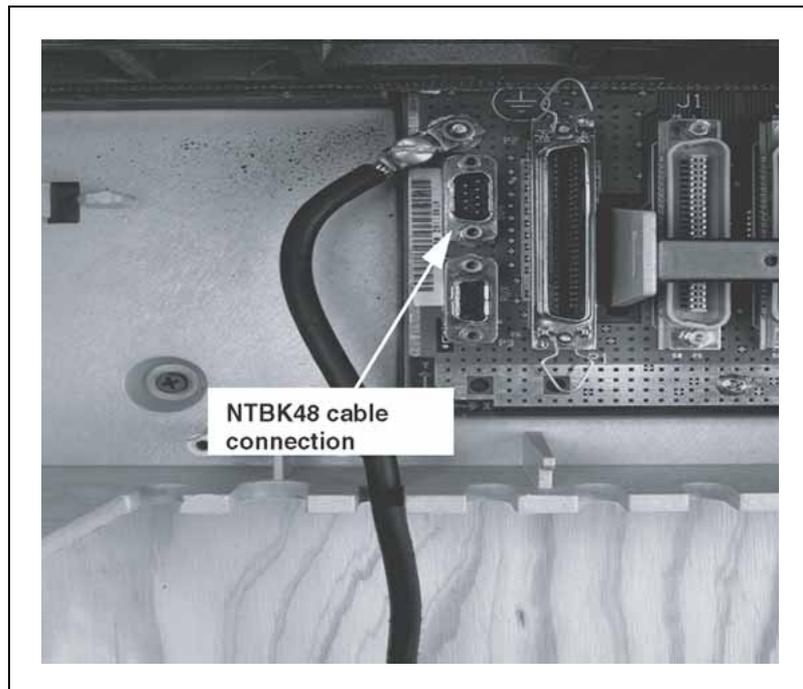
For the Chassis system, the SDI network interface is located on the bottom left side at the rear of the chassis (see).

The baud rate for port 0 is selected by setting switches on the faceplate of the SSC card. Baud rates for ports 1 and 2 are set using overlay programs.

Figure 29
Chassis SDI cable connector
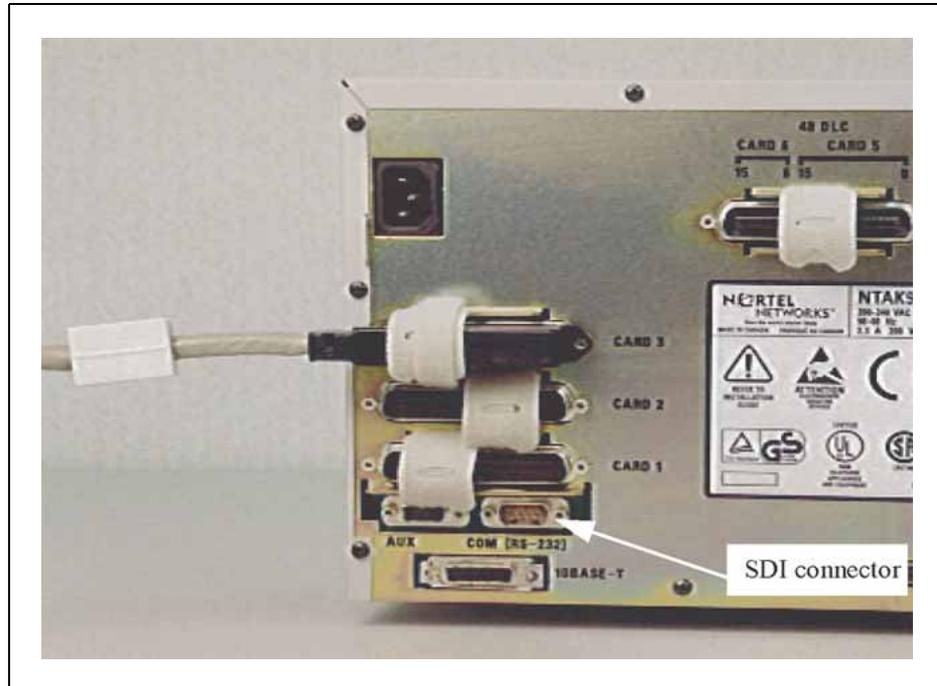
The baud rates available on all three ports are 300, 600, 1200, 2400, 4800, and 19200 baud.

Table 28
Default port configuration

TTY Number	Card	Port	Use	Configuration
0	0	0	MTC/SCH/BUG	1200/8/1/NONE
1	0	1	MTC/SCH/BUG	1200/8/1/NONE
2	0	2	CTY	1200/8/1/NONE

NTAK02 SDI/DCH card

The optional SDI/DCH card provides a maximum of four serial Input/Output (I/O) ports, which are grouped into two pairs:

* port 0 and port 1

• port 2 and port 3

Ports 1 and 3 may be configured as DCH or ESDI. Ports 0 and 2 may only be configured as SDI. Each pair is controlled by a switch, as shown in .

Table 29
Switch settings

Port 0	Port 1	SW 1-1	SW 1-2
SDI	DCH/DPNSS	OFF	OFF
SDI	DCH/DPNSS	OFF	ON
—	ESDI	ON	ON

Port 2	Port 3	SW 1-3	SW 1-4
SDI	DCH/DPNSS	OFF	OFF
SDI	DCH/DPNSS	OFF	ON
—	ESDI	ON	ON

In the UK, Digital Private Network Signaling System (DPNSS) can replace the DCH function.

Two ports offer the option for DTE/DCE configuration. This option is selected from a jumper on the card. shows the jumper settings:

Table 30
Jumper settings

Port	Jumper location	Strap for DTE	Strap for DCE	Jumper location	RS422	RS232
0	J10	C - B	B - A			
1	J7	C - B	B - A	J9	C - B	B - A
	J6	C - B	B - A	J8	C - B	B - A
2	J5	C - B	B - A			
3	J4	C - B	B - A	J2	C - B	B - A
	J3	C - B	B - A	J1	C - B	B - A

Connecting to the ports

The methods by which external devices can be connected to the SDI/DCH card are:

- Use the NTAK19FB four-port SDI cable. This cable does not have to be terminated at the cross-connect terminal since it is equipped with connectors.

- Use the NE-A25-B cable and terminate it at the cross-connect terminal. Table 31 "NTAK02 pinouts Port 0 at the cross-connect terminal" (page 123) through Table 34 "NTAK02 connections at the cross-connect terminal Port 3" (page 125) give the pinouts for the SDI/DCH card.

Table 31
NTAK02 pinouts Port 0 at the cross-connect terminal

RS232					
Cable		Signal		Designations I = input O = output	
Pair	Color	DTE	DCE	DTE	DCE
1T	W-BL	0	0	-	-
1R	BL-W	DTR	DCD	O	I
2T	W-O	DSR	CH/CI	I	O
2R	O-W	DCD	DTR	I	O
3T	W-G	RTS	CTS	O	I
3R	G-W	CTS	RTS	I	O
4T	W-BR	RX	TX	I	O
4R	BR-W	TX	RX	O	I
5T	W-S	-	-	-	-
5R	S-W	SG	SG	-	-

Table 32
NTAK02 connections at the cross-connect terminal Port 1

RS422						RS232			
Cable		Signal		Designations I = input O = output		Designations I = input O = output		Signal	
Pair	Color	DTE	DCE	DTE	DCE	DTE	DCE	DTE	DCE
5T	W-S	SCTE A	SCTA	O	I	O	I	SCT	SCT
5R	S-W	-	-	-	-	-	-	-	-
6T	R-BL	SCTE B	SCTB	O	I	-	-	CH/CI	-
6R	BL-R	DTR	DCD	O	I	-	-	DTR	DCD
7T	R-O	DSR	CH/CI	I	O	I	O	DSR	CH/CI

RS422						RS232			
Cable		**Signal**		**Designations** **I = input** **O = output**		**Designations** **I = input** **O = output**		**Signal**	
Pair	**Color**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**
7R	O-R	DCD	DTR	I	O	I	O	DCD	DTR
8T 8R	R-G G-R	RTS CTS	CTS RTS	O I	I O	O I	I O	RTS CTS	CTS RTS
9T 9R	R-BR BR-R	SCRA SCTA	SCTE A RXCA	I I	O O	I I	O O	SCR SCT	SCT -
10T 10R	R-S S-R	SCRB SCTB	SCTE B RXCB	I I	O O	- -	- -	- -	- -
11T 11R	BK-BL BL-BK	RXDA TXDA	TXDA RXDA	I O	O I	I O	O I	RXD TXD	TXD RXD
12T 12R	BK-O O-BK	RXDB TXDB	TXDB RXDB	I O	O I	- -	- -	- -	- -
25T 25R	V-S S-V	SG -	SG -	- -	- -	- -	- -	SG -	SG -

Table 33
NTAK02 connections at the cross-connect terminal Port 2

RS422						RS232			
Cable		**Signal**		**Designations** **I = input** **O = output**		**Designations** **I = input** **O = output**		**Signal**	
Pair	**Color**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**
13T 13R	BK-G G-BK			- -	- -	- O	- I	- DTR	- DCD
14T 14R	BK-B R BR-B K			- -	- -	I I	O O	DSR DCD	CH/CI DTR
15T 15R	BK-S S-BK			- -	- -	O I	I O	RTS CTS	CTS RTS

RS422						RS232			
Cable		**Signal**		**Designations** **I = input** **O = output**		**Designations** **I = input** **O = output**		**Signal**	
Pair	**Color**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**
16T 16R	Y-BL BL-Y			- -	- -	I O	O I	RX TX	TXD RXD
17T 17R	Y-O O-Y			O -	I -	O -	I -	- SG	- SG

Table 34
NTAK02 connections at the cross-connect terminal Port 3

RS422						RS232			
Cable		**Signal**		**Designations** **I = input** **O = output**		**Designations** **I = input** **O = output**		**Signal**	
Pair	**Color**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**	**DTE**	**DCE**
17T 17R	Y-O O-Y	SCTE A -	SCTA -	O -	I -	O -	I -	SCT -	SCT -
18T 18R	Y-G G-Y	SCTE B DTR	SCTB DCD	O O	I I	- -	- -	CH/CI DTR	- DCD
19T 19R	Y-BR BR-Y	DSR DCD	CH/CI DTR	I I	O O	I I	O O	DSR DCD	CH/CI DTR
20T 20R	Y-S S-Y	RTS CTS	CTS RTS	O I	I O	O I	I O	RTS CTS	CTS RTS
21T 21R	V-BL BL-V	SCRA SCTA	SCTE A RXCA	I I	O O	I I	O O	SCR SCT	SCT -
22T 22R	V-O O-V	SCRB SCTB	SCTE B RXCB	I I	O O	- -	- -	- -	- -
23T 23R	V-G G-V	RXDA TXDA	TXDA RXDA	I O	O I	I O	O I	RXD TXD	TXD RXD
24T 24R	V-BR BR-V	RXDB TXDB	TXDB RXDB	I O	O I	- -	- -	- -	- -
25T 25R	V-S S-V	- SG	- SG	- -	- -	- -	- -	SG -	SG -

Characteristics of the low-speed port

Ports 0 and 2 are asynchronous, low-speed ports. They transfer data to and from the line one bit at a time.

The characteristics of the low-speed port are as follows:

- **Baud rate**: 300; 600; 1200; 2400; 4800; 9600; 19,200
  Default 1200.

- **Parity**: Odd, even, none.
  Default none.

- **Stop bits**: 1, 1.5, 2
  Default 1

- **Flow control**: XON/XOFF, CTS, none.
  Default none.

- **Duplex**: Full.

- **Interface**: RS-232-D

- **Data bits**: 5, 6, 7, 8
  Default 8.

Characteristics of the high-speed port

Ports 1 and 3 are synchronous, high-speed ports with the following characteristics:

- **Baud rate**: 1200; 2400; 4800; 9600; 19,200; 56,000; 64,000.

- **Data bit**: Transparent (1).

- **Duplex**: Full.

- **Clock**: Internal or external.

- **Interface**: RS-232-D, RS-422-A.

ESDI settings

Port 9 is preprogrammed as an ESDI network interface and supports Meridian Mail and Call Pilot. It functions as a Command Status Link with settings as shown in Table 35 "ESDI settings" (page 126).

Table 35
ESDI settings

Setting	Code
ESDI	YES
SYNC	YES
DUPX	FULL
BPS	4800

Setting	Code
CLOK	EXT
IADR	003
RADR	001
T1	10
T2	002
T3	040
N1	128
N2	08
K	7
RXMT	05
CRC	10
ORUR	005
ABOR	005
USER	CMS

NTDK23, NTDK25, and NTDK80 Fiber Receiver cards

The NTDK23, NTDK25, and NTDK80 Receiver cards used in Small System support one SDI network interface.

Parameter settings

Baud rates are selected by setting switches located in the faceplate of each Fiber Receiver card. The available settings are:

- 150, 300, 600, 1200, 2400, 4800, 9600 and 19200 baud

Other RS232 parameters are fixed as shown in .

Table 36
Fixed parameter settings

Parameter	Setting
Parity	None
Mode	Asynchronous
Stop Bits	1
Data Bits	8

The port can be used for MTC/SCH/BUG modes.

Connection to external equipment

The connection to external devices (such as TTYs, modems and so on) is achieved through the nine-pin SDI connector located in the Expansion Cabinet. It is extended to the external equipment with an NTAK1118 single-port SDI cable.

Card slot and loop assignments

Contents

This section contains information on the following topics:

Developing a card slot assignment plan

A card slot allocation plan showing circuit card-to-slot assignments should be prepared in advance for each cabinet. See the most current Small System product bulletins for minimum vintage requirements.

First allocate to the Main Cabinet or Chassis the cards that must reside there. Fill in the remaining card slots as required.

Card slot assignments for the Cabinet system

The NTDK20 Small System Controller (SSC) card must be installed in the SSC slot (slot 0) of the Main and IP Expansion Cabinets. Refer to Table 21 "System controller card requirements for Small Systems" (page 96) for SSC card requirements.

> *Note:* When the NT6D70 card is used as a clock controller, it MUST be installed in the Main Cabinet slots 1-9.

If you plan on using the preassigned numbering plan with consecutive numbers, it is important to assign all line cards in consecutive card slots.

To prepare the card slot assignment plan, complete Table 37 "Card slot assignment plan for the Cabinet system" (page 130) to list the total number of circuit cards required for the installation.

Table 37
Card slot assignment plan for the Cabinet system

Card	Card slot	Number of cards
Used only in the Main Cabinet		
NT6D70 SILC (if clock controller is active — see Note 1)		
CallPilot		
NTBK22 MISP		
Used only in fiber Expansion Cabinets		
NTDK23 10 m Fiber Receiver card (see Note 2)		
NTDK25 or NTDK80 3 km Fiber Receiver card (see Note 2)		
Used in the Main and Expansion Cabinets		
NTDK20 SSC (one per Main and one per IP Expansion Cabinet — see Note 3)	0	
NTAK02 SDI/DCH		
NTAK79 2.0 Mb PRI		
NTDK10 2.0 Mb DTI		
NTRB21 1.5 Mb TMDI		
NTDK50 2.0 Mb PRI		
NT8D02 Digital line card		
NT8D03 Analog line card		
NT8D09 Message waiting		
NT8D14 Universal Trunk		
NT8D16 Digitone Receiver		
NT8D15 E&M Trunk		

Note 1: The NT6D70 SILC card must be installed in the Main Cabinet (slots 1 through 9) if it is used as a clock controller.

Note 2: Each fiber Expansion Cabinet must have either an NTDK23, NTDK25, or NTDK80 Fiber Receiver card positioned in slot 0

Note 3: The NTDK20 SSC card must be installed in slot 0 in the Main and IP Expansion Cabinets. Refer to Table 21 "System controller card requirements for Small Systems" (page 96) for SSC card requirements.

Card	Card slot	Number of cards
NT7D16 Data Access		
NT6D70 SILC (see Note 1)		
NT6D71 UILC		
NT5K02 XFALC		
NT5K18 XFCOT		
NT5K17 XDDI		
NT5K19 XFEM		
NT5K36 XDID/DOD		
NT5K21 XMFC/MFE		
NTAG26 XMFR		

Note 1: The NT6D70 SILC card must be installed in the Main Cabinet (slots 1 through 9) if it is used as a clock controller.

Note 2: Each fiber Expansion Cabinet must have either an NTDK23, NTDK25, or NTDK80 Fiber Receiver card positioned in slot 0

Note 3: The NTDK20 SSC card must be installed in slot 0 in the Main and IP Expansion Cabinets. Refer to Table 21 "System controller card requirements for Small Systems" (page 96) for SSC card requirements.

WARNING

If NE-A25B cables are used instead of NTAK19AA and NTAK19BA cables with the NTAK02 card, proceed with care. NE-A25B cables are not wired out to station equipment or trunk circuits. They can be wired out only to SDI circuits.

Figure 30 "Main Cabinet card slot assignments" (page 132) shows the card slot numbers and assignments in the Main Cabinet.

Figure 30
Main Cabinet card slot assignments

IPE, SDI/DCH, or Meridian Mail
(12-port Meridian Mail also requires card slot 9)

IPE, TDS/DTR, SDI/DCH, DTI/PRI, MISP
(Note: On EMC systems, active CCs must be in slots 1-3)

SSC Card

Power Supply

N T D K 2 0

Meridian Mail

Pwr Sup | 0 CPU | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 10 MM1 | MM2 | MM3

553-AAA1562.EPS
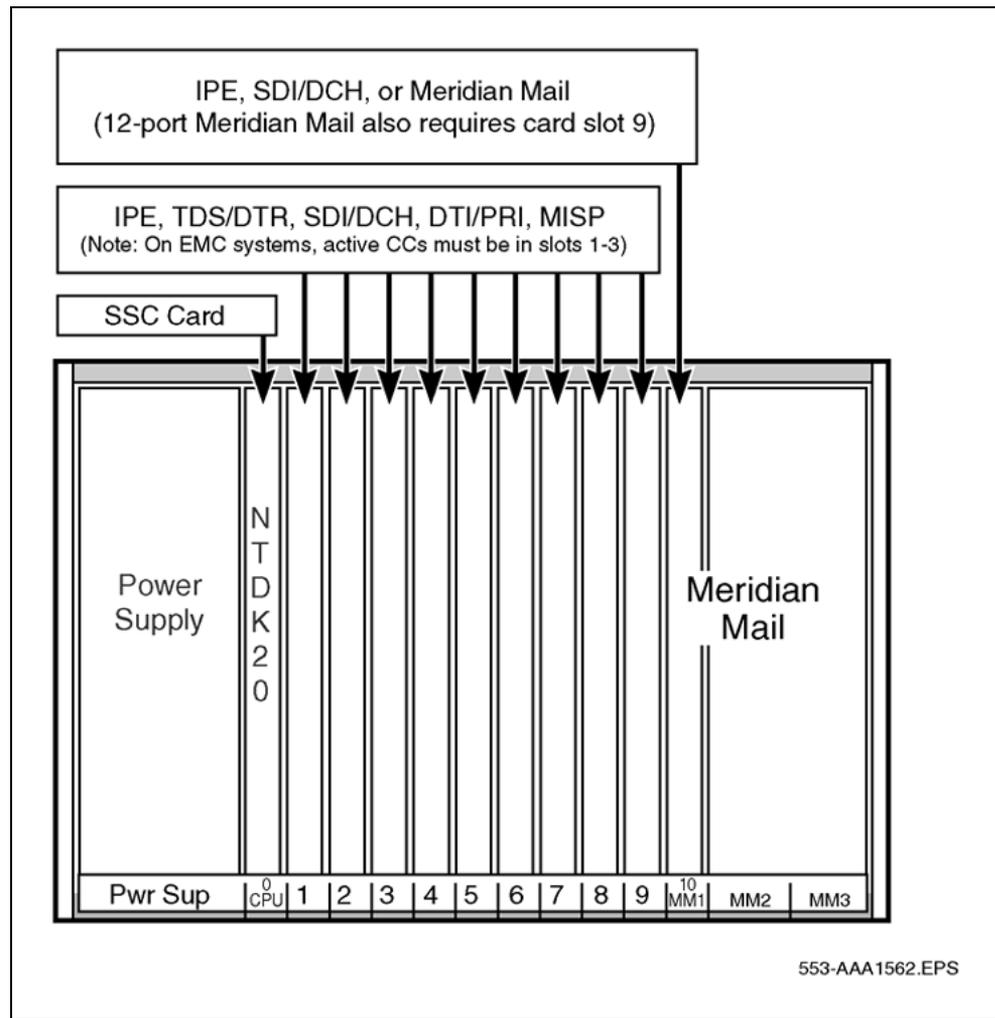
IPE card slot assignments with IP expansion

If you are using IP Expansion Cabinets or Chassis, then trunk and line cards can be distributed throughout each of the system cabinets or chassis in such a way as to allow for survivable operation. The intent is for a cabinet or chassis equipped with both trunk and line cards in survival mode to still handle calls.

IPE card slot assignments: non-IP expansion

If you are not using IP Expansion, then place trunk and line cards in the system cabinets in such a way as to allow for future expansion. Line cards are placed in the left-hand slots of the cabinets. If the system is using the default numbering plan and consecutive DN numbering is desired, place the line cards one after another, leaving no blank slots in between. Trunk

cards are placed in the right-hand slots of the cabinets. Figure 31 "Card slot assignment plan: one-cabinet system" (page 133) shows the card slot assignment plan for a one-cabinet system

Plan the card slot assignments so that the trunk and line card growth is towards the middle. For example, Figure 32 "Card slot assignment plan: two-cabinet system without IP expansion" (page 134) shows the slot assignment plan for systems equipped with two Expansion Cabinets. Figure 33 "Card slot assignment plan: three-cabinet system without IP expansion" (page 134) and Figure 34 "Card slot assignment plan: four-cabinet system without IP expansion" (page 134) and Figure 35 "Card slot assignment plan: five-cabinet system without IP expansion" (page 135) on page Figure 34 "Card slot assignment plan: four-cabinet system without IP expansion" (page 134)show the slot assignment plan for systems equipped with three, four, and five Expansion Cabinets, respectively.

Figure 31
Card slot assignment plan: one-cabinet system
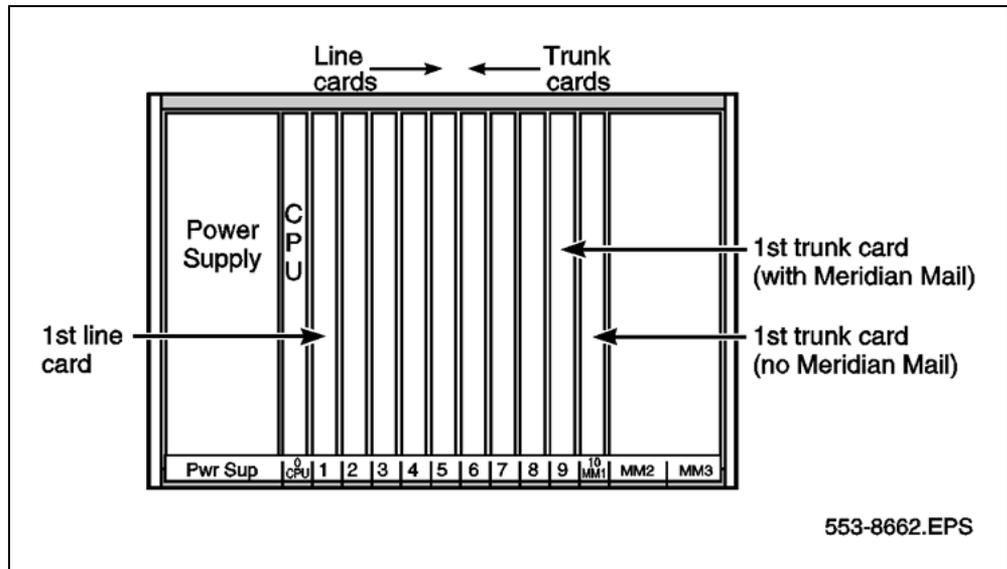
Figure 32
Card slot assignment plan: two-cabinet system without IP expansion
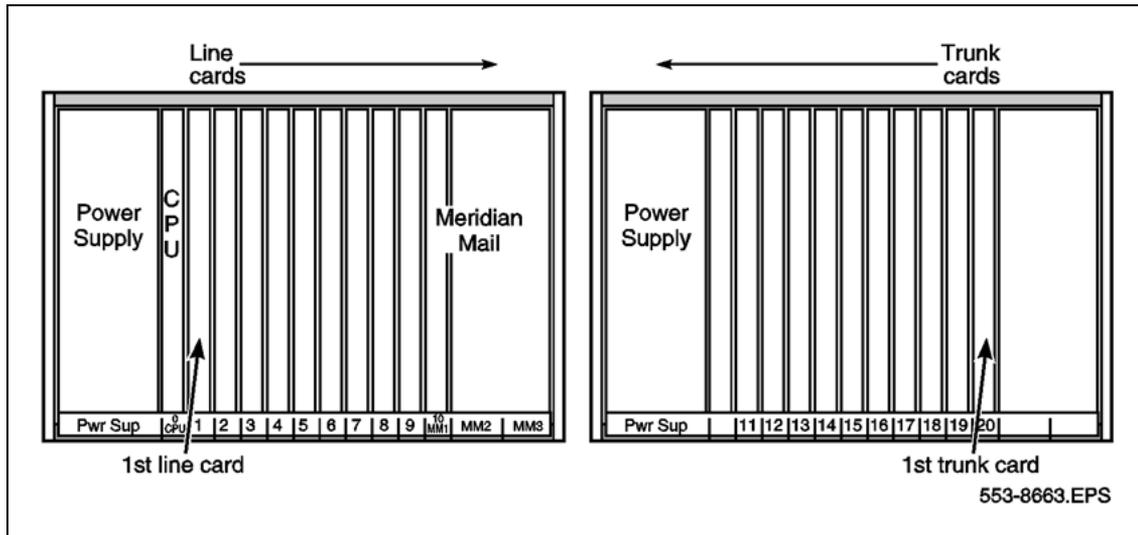
Figure 33
Card slot assignment plan: three-cabinet system without IP expansion
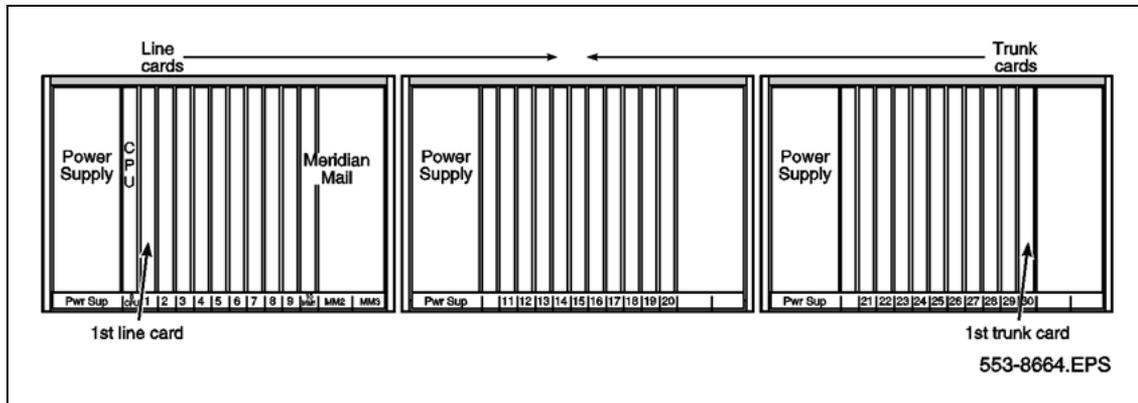
Figure 34
Card slot assignment plan: four-cabinet system without IP expansion
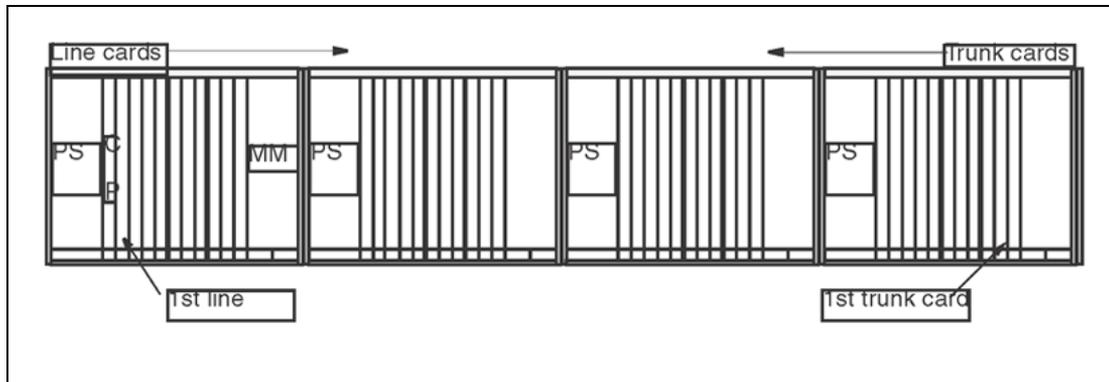
Figure 35
Card slot assignment plan: five-cabinet system without IP expansion
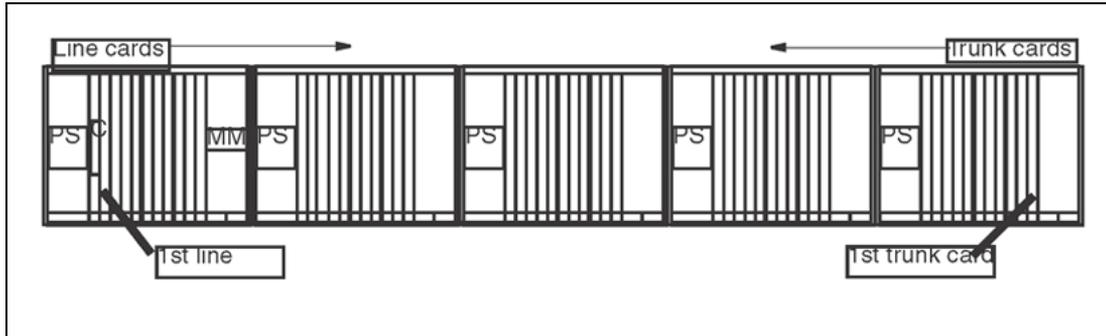
Card slot assignments for the Chassis system

Prepare a card slot assignment plan in advance. The card slot allocation plan shows circuit card-to-slot assignments. See the most current Small System product bulletins for minimum version requirements.

Note: You must insert the NTDK20 Small System Controller (SSC) card in Slot 0 of the chassis and any IP Expansion Chassis.

Slot 4 accepts the NTDK16 48-port Digital Line Card (DLC) only. However, you can place a card that takes two slots in slot 3, and it can overlap into slot 4.

When expanding older Option 11C Mini systems using expansion daughterboards, the NTDK97 Mini System Controller (MSC) card in the chassis must be replaced by the NTDK20 SSC card. Refer to "Expansion daughterboards" (page 100) for information on the expansion daughterboards that the SSC card supports.

You can install the following cards in slots 1, 2, and 3 of the chassis:

- NTAK10

- NTBK50

- NTAK79

- NTBK22

- NT6D70 (when used as a clock controller)

- NTRB21

The NTBK50 supports the following optional daughterboards:

- NTAK20 Clock Controller

- NTBK51 DDCH daughterboard or the NTAK93 D-channel Interface

To prepare a plan for card slot assignment, write the total number of circuit cards required for the installation in .

Table 38
Card slot assignment plan

Card	Card slot	Number of cards
Used only in the chassis		
NTDK20 SSC	0 only	1
NTDK16 48-port DLC	4 only	
NTAK02 SDI/DCH		
NTRB21 1.5 Mbit DTI/PRI		
NTAK10 2.0 Mbit DTI		
NTAK79 2.0 Mbit PRI		
NTBK50 2.0 Mbit PRI		
NT5K20 Tone Detector		
NT5K48 Tone Detector		
NTBK22 MISP		
NT6D70 SILC		
Used only in the chassis expander		
NT6R16 Meridian Mail Mini	10 only	
Used only in the fiber Expansion Chassis		
NTDK23 10 m Fiber Receiver card		
NTDK25 or NTDK80 3 km Fiber Receiver card		
Used in the chassis and the chassis expander		
NT8D02 Digital Line Card		
NT8D09 Message Waiting		
NT8D14 Universal Trunk		
NT8D16 Digitone Receiver		
NT8D15 E&M Trunk		
NT7D16 Data Access		
NT6D70 SILC (see Note)		
NT6D71 UILC		
NT5K02 XFALC		
Note: Install the NT6D70 SILC card in the chassis (Slots 1, 2, or 3) if it is used as a clock controller.		

Card	Card slot	Number of cards
NT5K18 XFCOT		
NT5K17 XDDI		
NT5K19 XFEM		
NT5K36 XDID/DOD		
NT5K21 XMFC/MFE		
NTAG26 XMFR		
Note: Install the NT6D70 SILC card in the chassis (Slots 1, 2, or 3) if it is used as a clock controller.		

WARNING

If you use NE-A25B cables instead of NTAK19AA and NTAK19BA cables with the NTAK02 card, continue with caution. NE-A25B cables are not wired out to station equipment or trunk circuits. NE-A25B cables can be wired out only to SDI circuits.

For expanded Chassis systems, the SSC card must be provisioned with either fiber Expansion or IP Expansion daughterboards.

Each fiber Expansion Chassis must have an NTDK23, NTDK25, or NTDK80 Fiber Receiver card positioned in slot 0.

Each IP Expansion Chassis must have an SSC card provisioned with an IP Expansion daughterboard.

Make sure to first allocate the cards that you must install in the chassis. Fill the remaining card slots as required.

If you plan on using the preassigned numbering plan with consecutive numbers, assign all line cards in consecutive card slots.

See and for the card slot assignments in the chassis and chassis expander.

IPE card slot assignments on chassis without IP expansion
Digital trunks cards must be placed in the chassis. Slot 4 must contain the 48-port DLC. shows the typical card slot assignment for a chassis.

Note: Slot 4 is keyed to prevent accidental insertion of cards other than the 48-port DLC.

IPE card slot assignments on the chassis expander
Any IPE card may be placed in cards slots 7 through 9. Slot 10 can contain any IPE card or the Meridian Mail. Refer to Figure 37 "Card slot assignments for the chassis expander" (page 139).

When planning the number of card slots that will be required in a system, the following items must be considered in addition to IPE card requirements:

- Additional SDI/DCHI/ESDI ports

- Tone Detectors (International only)

- Adding Meridian Mail

Figure 36
Card slot assignments for the chassis
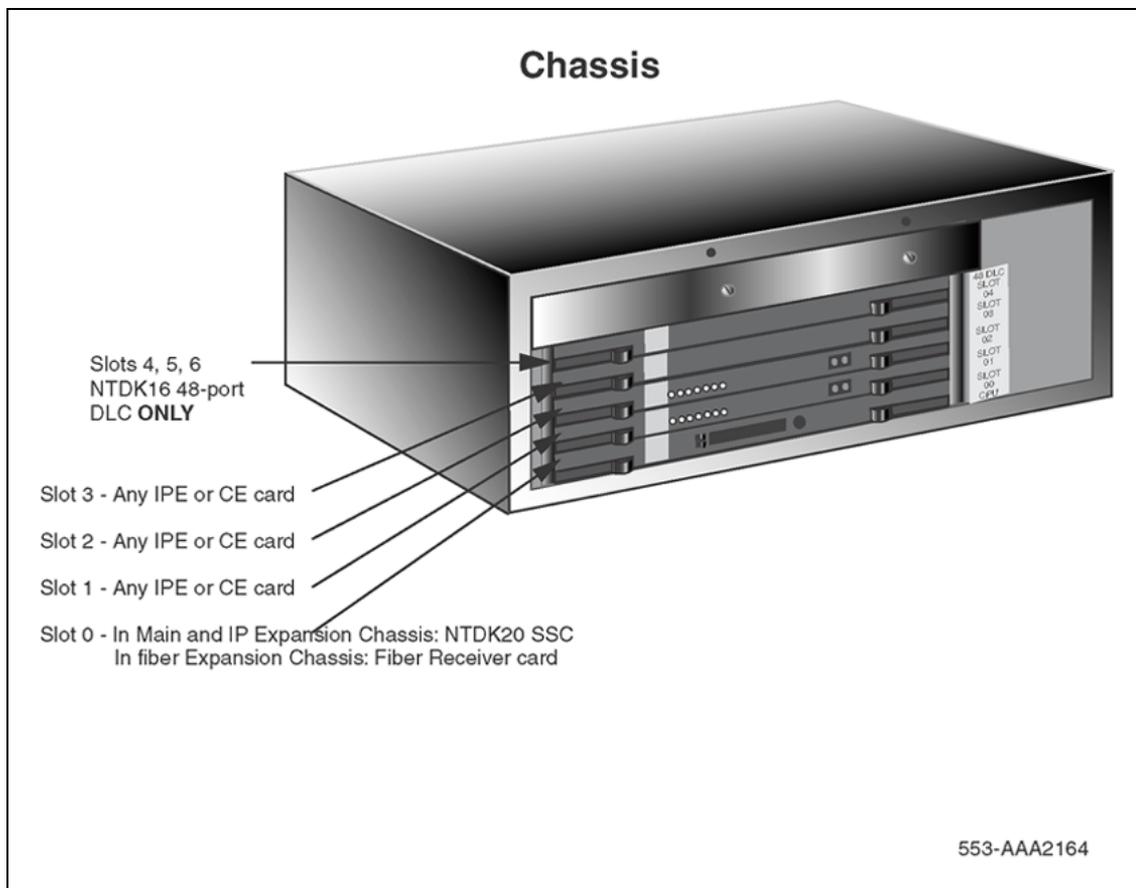
Note 1: Refer to Table 38 "Card slot assignment plan" (page 136) for a list of cards that you can insert in the chassis.

Note 2: Slot 4 accepts the NTDK16 48-port DLC card only. However, you can place a card that takes two slots in slot 3, and it can overlap into slot 4.

Figure 37
Card slot assignments for the chassis expander
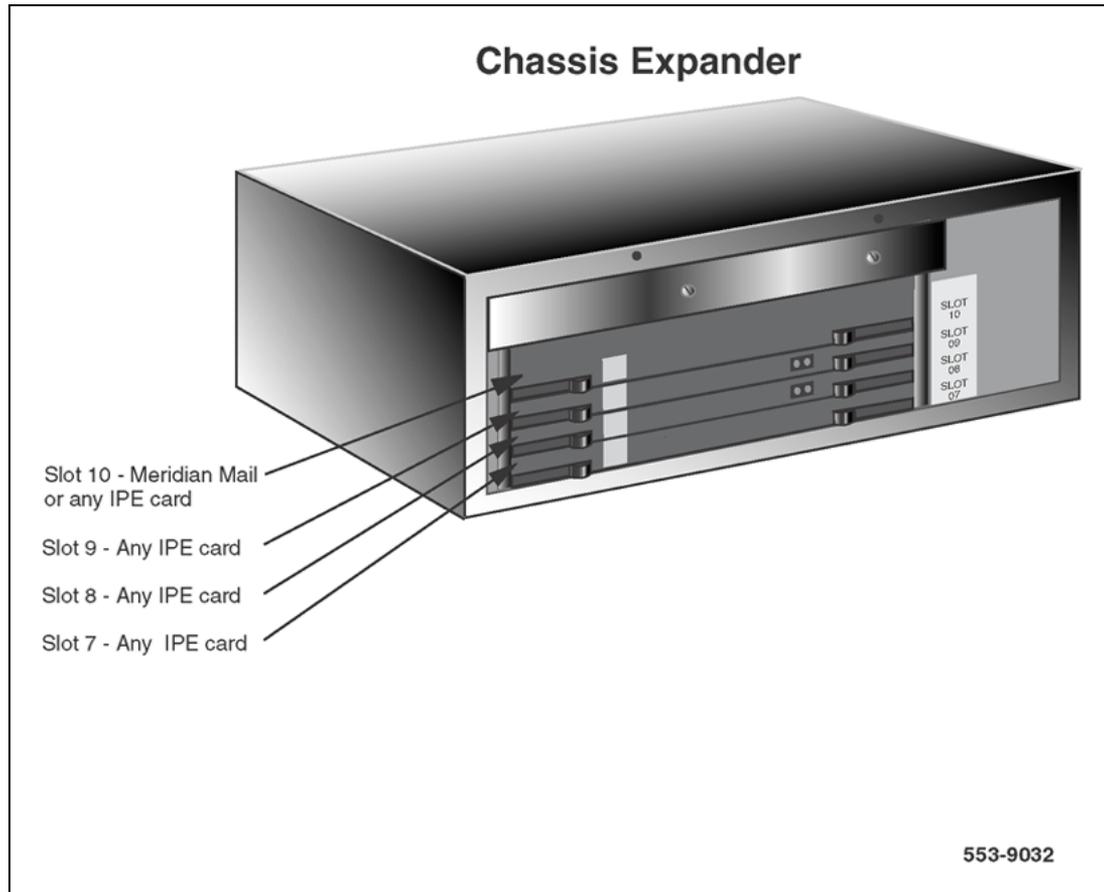
> *Note:* Refer to Table 38 "Card slot assignment plan" (page 136) for a list
> of cards that you can insert in the chassis expander.

Loops and superloops

Each IPE circuit card has a loop entirely dedicated to it. Every group of four
IPE card slots is programmed as an individual superloop. These superloops
are configured by default and are not programmed by the user.

There are a total of 640 timeslots (channels) for each Small System. Each
superloop provides 120 timeslots, while an IPE slot provides 30 timeslots.

Table 39 "Small System superloops — fiber expansion" (page 140) details the superloop configuration for Cabinet or Chassis systems with fiber expansion.

Table 39
Small System superloops — fiber expansion

Main			Expansion 1			Expansion 2			Expansion 3			Expansion 4		
CS	CL	SL	CS	CL	SL	CS	CL	SL	CS	CL	SL	CS	CL	SL
1	20	0	11	—	8	21	—	32	31	—	40	41	—	64
2	21	0	12		8	22		32	32		40	42		64
3	22	0	13		12	23		32	33		44	43		64
4	23	0	14		12	24		32	34		44	44		64
5	24	4	15		12	25		36	35		44	45		68
6	25	4	16		12	26		36	36		44	46		68
7	26	4	17		16	27		36	37		48	47		68
8	27	4	18		16	28		36	38		48	48		68
9	28	8	19		16	29		40	39		48	49		72
10		8	20		16	30		40	40		48	50		72
CS = Card Slot, CL = CE Loop, SL = Superloop														

Table 40 "Small System superloops — IP expansion" (page 140) details the superloop configuration for Cabinet or Chassis systems with IP expansion.

Table 40
Small System superloops — IP expansion

Main			Expansion 1			Expansion 2			Expansion 3			Expansion 4		
CS	CL	SL	CS	CL	SL	CS	CL	SL	CS	CL	SL	CS	CL	SL
1	20	0	11	52	8	21	76	32	31	85	40	41	116	64
2	21	0	12	53	8	22	77	32	32	86	40	42	117	64
3	22	0	13	54	12	23	78	32	33	87	44	43	118	64
4	23	0	14	55	12	24	79	32	34	88	44	44	119	64
5	24	4	15	56	12	25	80	36	35	89	44	45	120	68
CS = Card Slot, CL = CE Loop, SL = Superloop														

Main			Expansion 1			Expansion 2			Expansion 3			Expansion 4		
CS	**CL**	**SL**	**CS**	**CL**	**SL**	**CS**	**CL**	**SL**	**CS**	**CL**	**SL**	**CS**	**CL**	**SL**
6	25	4	16	57	12	26	81	36	36	90	44	46	121	68
7	26	4	17	58	16	27	82	36	37	91	48	47	122	68
8	27	4	18	59	16	28	83	36	38	92	48	48	123	68
9	28	8	19	60	16	29	84	40	39	93	48	49	124	72
10		8	20		16	30		40	40		48	50		72
CS = Card Slot, CL = CE Loop, SL = Superloop														

Phantom and virtual loops

There are five user-programmable superloops available for use by either phantom or virtual TNs. Phantom loops support a maximum of 624 TNs. Virtual loops support a maximum of 1248 TNs.

A superloop cannot have both phantom and virtual sets defined on it. A phantom loop has only phantom TNs and a virtual loop has only virtual TNs.

Virtual superloops can be configured with up to 32 units per card, while phantom loops are limited to 16 units per card.

The range of phantom and virtual loops is 96–112, grouped by fours into phantom and virtual superloops. Table 41 "Phantom and virtual superloops" (page 141) shows how the phantom or virtual superloops map into cards to configure the phantom or virtual TNs.

Note: The order of the columns in Table 41 "Phantom and virtual superloops" (page 141) is significant: For virtual and phantom TNs, the user must define the superloop first and then define the cards.

Table 41
Phantom and virtual superloops

SL	CS	SL	CS	SL	CS	SL	CS	SL	CS
96	61	100	65	104	69	108	73	112	77
96	62	100	66	104	70	108	74	112	78
96	63	100	67	104	71	108	75	112	79
96	64	100	68	104	72	108	76	112	80
SL = Superloop, CS = Card Slot									

SL	CS	SL	CS	SL	CS	SL	CS	SL	CS
96	81	100	85	104	89	108	93	112	97
96	82	100	86	104	90	108	94	112	98
96	83	100	87	104	91	108	95	112	99
96	84	100	88	104	92	108	96	112	n/a
SL = Superloop, CS = Card Slot									

For more information on phantom and virtual loops, refer to Features and Services (553-3001-306).

Design parameters

Contents

This section contains information on the following topics:

Introduction

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

System parameters

Table 42 "System parameters" (page 144) lists system parameters and provides their maximum values.

Table 42
System parameters

System parameters	Maximum value	Comments
Customers	100	
Display messages for background terminal	255	
Input/output ports (for example, TTYs, printers)	16	Each MSDL counts as one device; a history file counts as one device.
TNs	1744	Software design limit. Actual number of TNs will be constrained by physical capacity, real time, memory, and License limits.

Customer parameters

Table 43 "Customer parameters" (page 144) lists customer parameters and their maximum values.

Table 43
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 44 "Console and telephone related parameters" (page 145) lists console and telephone-related parameters and their maximum values.

Table 44
Console and telephone related parameters

Console/telephone parameters	Maximum value	Comments
Consoles per customer	63	
Lamp field arrays per customer	1	May be repeated once on another console.
Lamps per array (all numbers must be consecutive)	150	
Feature keys per attendant console: – M2250	 20	
Incoming call indicators per console	20	
Trunk group busy indicators per console: – M2250	 20	
Additional key/lamp strips: – console – telephones	 2 6	
Add on modules: – M3904 Key Expansion Module (KEM) – IP Phone 2002 KEM – IP Phone 2004 KEM	 2 1 one-page KEM 2 one-page KEM or 1 two-page KEM	
Protect bcs block length	512	
Associate sets (ASTs)	1000	

Trunk and route parameters

Table 45 "Trunk and network-related parameters" (page 146) lists trunk and network-related parameters and their maximum values.

Table 45
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 4000	1000 without ESN Location Code Expansion package (400), 4000 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 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 46 "ACD feature parameters" (page 147) lists ACD feature parameters and their maximum values.

Table 46
ACD feature parameters

ACD parameters	Maximum value	Comments
ACD DNs and CDNs per customer	1000	The ACDE package is required, otherwise the limit is 240.
Agent positions per DN	1200	Real-time and physical capacity constraints can limit this further.
Agent priorities	48	
Agent IDs per customer	9999	
Agents logged in at one time per system	9999	Real-time constraints can limit this further.
AST DNs per telephone	2	
Number of ACD-ADS customers	5	
Terminals and printers on CCR	8	
Links per VASID	1	

Special feature parameters

Table 47 "Non-ACD feature parameters" (page 147) lists non-ACD feature parameters and their maximum values.

Table 47
Non-ACD feature parameters

Feature parameters	Maximum value	Comments
Speed call lists per system	8191	The number of speed call lists and the number of DNs per speed call list can be limited by the amount of available memory on the system (protected and unprotected data store).
Number of DNs in speed call list	1000	
Multiple appearances of the same directory number (MADN)	30*	Limited by watchdog timer. *See Steps in a hunting group.

Feature parameters	Maximum value	Comments
Steps in a hunting group	30*	Marketing objective, limited by watchdog timer. *In combination with MADN, each hunt step with more than 16 appearances is counted as two, so the maximum combination of MADN and hunt steps is 30 MADN and 15 hunt steps.
Number of Call Party Name Display names defined	Variable	Limited by the number of DNs defined and available space in the protected data store.
CPND length: – SL-1 protocol – ISDN protocol	 27 24	– Software design limit. – Display IE limitation (DMS switches have a display IE limit of 15).
AWU calls in 5 minutes	500	Marketing objective, constrained by ring generator.
Group Call Feature: – Groups per customer – Stations per group	 64 10	
BRI application: – Protocol parameter 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 the number of DSLs, memory, and real time.

Hardware and capacity parameters

The software design limits are not typically the binding constraints. The number of items of a particular type is usually determined by a combination of loop and slot constraints (if the item requires loops) or by slot constraints alone.

Table 48 "Physical capacity/hardware-related parameters" (page 149) lists hardware and capacity parameters and their maximum values.

Table 48
Physical capacity/hardware-related parameters

Physical capacity/hardware parameters	Maximum value (loops)	Comments
Multifrequency sender cards	64	Software design limit; each MFS card requires 2 loops.
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		Each superloop requires 4 loops.
Digitone receivers	255	Software design limit.
Multifrequency receivers	255	Software design limit.
Tone detectors	255	Software design limit.

Voice Gateway Media Cards

A Voice Gateway Media Card is any Media Card running the IP Line application.

A Voice Gateway Media Card has 32 ports. Its transcoding traffic comes from trunk or line cards. Voice Gateway Media Cards can be assigned to any slot other than slot 0. To minimize blocking at the superloop, assign no more than four 32-port Media Cards per superloop.

Memory-related parameters

Table 49 "Memory-related parameters" (page 150) lists memory-related parameters and their maximum values.

Table 49
Memory-related parameters

Parameter	Values
Low-priority input buffers	95 − 1000
• (recommended default)	(450)
High-priority input buffers	16 − 1000
• (recommended default)	(450)
Input buffer size (words)	4
500-set, trunk and digital set output buffers	16 − 5000
• (recommended default)	(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	26 − 2047
• recommended default	800
Call registers assigned to AUX	26–255

Note 1: In a system with Meridian Mail, CallPilot, AML, and Symposium, add the number of CSQI and CSQO to the Call Register (CR) requirement obtained from feature impact calculations.

Note 2: The buffer estimates were based on relatively conservative scenarios, which should cover most practical applications in the field. However, most models deal with "average traffic". When traffic spikes occur, buffers can overflow. In these cases, raise the buffer size, depending on the availability of CRs. The maximum number of buffers allowed for CSQI and CSQO is 25% of NCR or 4095, whichever is less.

Parameter	Values
Number of AML msg call registers	25 – the minimum of 25% of total call registers or 255 (default 25)
Call registers for CSL input queues (CSQI)	255 – the minimum of 25% of total call registers or 4095 (default 255)
Call registers for CSL/AML output queues (CSQO)	255 – the minimum of 25% of total call registers or 4095 (default 255)
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 – 64 000

Note 1: In a system with Meridian Mail, CallPilot, AML, and Symposium, add the number of CSQI and CSQO to the Call Register (CR) requirement obtained from feature impact calculations.

Note 2: The buffer estimates were based on relatively conservative scenarios, which should cover most practical applications in the field. However, most models deal with "average traffic". When traffic spikes occur, buffers can overflow. In these cases, raise the buffer size, depending on the availability of CRs. The maximum number of buffers allowed for CSQI and CSQO is 25% of NCR or 4095, whichever is less.

Buffer limits

The buffer limit is the maximum number of 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 is not 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 49 "Memory-related parameters" (page 150) 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 49 "Memory-related parameters" (page 150). 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 Small System Call Center (maximum 25 000 CRs) using many applications (such as CallPilot), it is advisable to set the CSQI/CSQO to a high value (even up to the limit of 4095).

System capacities

Contents

This section contains information on the following topics:

Introduction

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" (page 205) provides the algorithms for engineering the system within the capacity limits. In some cases, applications such as Call Center require detailed engineering. These applications are discussed in "Application engineering" (page 249)

The worksheets in Appendix "Worksheets" (page 301)implement the algorithms.

Memory size

The Small System Controller (SSC) has a separate Flash EPROM memory for program store and DRAM for data store. Flash EPROM and the primary flash drive (C: drive) reside on the same flash daughterboard. The SSC has one SIMM slot for DRAM. It supports a maximum of 32 MByte of DRAM. Refer to "System controller cards and software daughterboards" (page 91) for more information about the SSC card.

Table 50 "CS 1000 Release 4.5 memory requirements" (page 154) shows the minimum amount of memory required for CS 1000 Release 4.5 software.

Table 50
CS 1000 Release 4.5 memory requirements

Processor	Flash memory required	DRAM memory required	Total memory
SSC	48 MByte	32 MByte	80 MByte

Table 51 "Recommended maximum call register counts" (page 154) gives the maximum call register count recommended for CS 1000 Release 4.5 software, so that the system's memory requirements do not exceed the processor's memory capacity.

Note: Sites experiencing memory shortages during an upgrade should check that the call register counts are within the bounds set by this table.

Table 51
Recommended maximum call register counts

System	Recommended call register count	Memory required (SL-1 words)	Memory required (MByte)
Cabinet system or Chassis system	800	181 600	0.693
Note: Call registers are 227 SL-1 words long. One SL-1 word is 4 bytes.			

Memory engineering

The data store consists of both protected and unprotected database information. Table 52 "Protected data store" (page 155) and Table 54 "Unprotected data store" (page 161) describe the information stored in each area and how to determine the values for input to "Worksheet 4: Protected memory calculations" (page 306) and "Worksheet 5: Unprotected memory calculations" (page 307).

The calculations described in Table 52 "Protected data store" (page 155) and Table 54 "Unprotected data store" (page 161) include references to memory store per item. Table 53 "PDS factors (units in SL-1 words)" (page 159) provides the memory store per item in protected data store. Table 55 "UDS factors (units in SL-1 words)" (page 165) provides the memory store per item in unprotected data store. These values are also referred to as the PDS factors and UDS factors, respectively.

The PDS and UDS factors in Tables Table 53 "PDS factors (units in SL-1 words)" (page 159) and Table 55 "UDS factors (units in SL-1 words)" (page 165) are based on assumptions about typical configurations, feature usage, and traffic patterns. The assumptions are specified in Tables Table 52 "Protected data store" (page 155) and Table 54 "Unprotected data store" (page 161), as they become relevant. Refer to Appendix "Worksheets" (page 301) for detailed calculations in cases where those assumptions may not apply.

Protected data store

Table 52 "Protected data store" (page 155) describes the protected data store (PDS) area.

Table 52
Protected data store

Item	Calculation*
Telephones: • 500/2500-type • ACD • M2006/2008 • 2216/2616 • M2317 • M3900 • IP Phones 200x • IP Softphones 2050 • Consoles • Add-on Modules • Templates • Attendants	Number of items × memory store per item
*Refer to Table 53 "PDS factors (units in SL-1 words)" (page 159) for the memory store per item (PDS factor).	

Item	Calculation*
Assumptions Average number of: • Features defined per 500/2500-type telephone = 8 • 500/2500-type telephones sharing the same template = 10 • Digital telephones sharing the same template = 2 • Non-key features per digital set = 4	
Data Service (DS)/VMS access TNs	(Number of CallPilot ports + Number of data-only ports) × memory store per DS/VMS access TN
Office Data Administration (ODAS)	(Number of CallPilot ports + Number of data ports only + Total number of sets + Number of analog trunks) × memory store for ODAS
Customers	Constant term + (Number of customers × memory store per customer)
Console Presentation Group (CPG) Level Services	Number of services × memory store per service
Directory Number (DN) translator **Assumptions** • The two lowest levels in the DN tree have average rate of 8 digits. • The rest of the DN tree has a structure that provides the lowest possible digit rate for upper levels.	(5.8 × Number of DNs) + 2 × (2 × Number of ACD DNs) + (Number of ACD positions + Number of DISA DNs) + (memory store per console × Number of consoles) + Number of dial intercom groups
Dial Intercom Group (DIG) translator	Maximum number of DIGs + 2 × (number of DIGs + Total number of the sets within DIGs)
Direct Inward System Access (DISA)	241 + (Number of DISA DNs × memory store per DISA DN)
*Refer to Table 53 "PDS factors (units in SL-1 words)" (page 159) for the memory store per item (PDS factor).	

Nortel Communication Server 1000
Communication Server 1000M and Meridian 1 Small System Planning and Engineering
NN43011-220   01.01   Standard
Release 5.0   30 May 2007

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Item	Calculation*
Authorization Code **Assumption** • The length of the authorization code is in the range of 4 through 7	(Number of customers × memory store per customer) + (1.47 × Number of authorization codes)
Speed Call	(Maximum number of Speed Call lists + Number of Speed Call lists) × (3 + 0.26 × Average number of entries per list × DN size)
Analog trunks	Number of analog trunks × memory store per analog trunk
Trunk Route	Constant term + (Number of trunk routes × memory store per trunk route) + (18 × each ISA route configured for any IFC)
Network	(Number of groups × memory store per group) + (Number of local loops × memory store per local loop) + (Number of remote loops × memory store per remote loop)
DTR, Tone Detector	(Number of DTRs × memory store per DTR) + (Number of TDETs × memory store per TDET)
Virtual Trunks	(Number of D-channels × memory store per D-channel) + (Number of Virtual Trunks × memory store per Virtual Trunk)
ISDN PRI/PRI2	(Number of D-channels × memory store per D-channel) + (Number of PRI trunks + Number of ISL trunks)
ISDN DTI/DTI2/JDMI	(Number of DTI loops × memory store per DTI loop) + (Number of DTI2 loops × memory store per DTI2 loop)
History file	Size for history file buffer
*Refer to for the memory store per item (PDS factor).	

Item	Calculation*
Basic Alternate Route Selection/Network Alternate Route Selection (BARS/NARS) **Assumptions** • The length of any code = 3 • The typical structure of the tree for every code (in terms of digit rate) is the following: — 10-10-10... for SPN code — 8 -10-10... for NXX/LOC code — 6-2-10-8-10... for NPA code	5684 + (31.21 × number of NPA Codes) + (1.06 × Number of NXX Codes) + (1.06 × (Number of LOC Codes) + (Number of SPN Codes) + (2 × Number of FCAS Tables)
ISDN Basic Rate Interface (BRI)	(Number of MISP boards × memory store per MISP board) + (Number of DSLs × memory store per DSL) + (Number of TSPs × memory store per TSP) + (Number of BRI DNs × memory store per BRI DN)
Coordinated Dialing Plan (CDP)	Constant term + (3 × Number of steering codes) + (8 × Number of route lists) + (3 × Total number of entries in route lists) + (3 × Number of distinct digit manipulation entries) + (Total number of digits inserted as part of digit manipulation ÷ 4)
Call Party Name Display (CPND)	Number of trunk routes + Number of consoles + Number of ACD DNs + Number of digital set DNs + Number of Names × (5 + Average length of name) + (Number of 1-digit DIG groups × 11) + (Number of 2-digit DIG groups × 101)
*Refer to for the memory store per item (PDS factor).	

Item	Calculation*
Feature Group D (FGD) Automatic Number Identification (ANI) Database **Assumptions** • All Numbering Plan Area (NPA) codes designated for BARS/NARS are used for ANI also. • One NPA block for every fifty NPA codes. • Five NXX blocks for each NPA block. • Twenty SUB blocks for each NXX block.	(3 × Number of NPA Codes) + (658 × Number of NPA codes)
Automatic Call Distribution (ACD)/Network ACD (NACD)	(Number of ACD DNs × memory store per ACD DN) + (Number of NACD DNs × memory store per NACD DN) + (Number of ACD positions × memory store per ACD position) + (Number of ACD agents) + (11 × Number of customers)
System overhead	Memory store for system overhead
*Refer to Table 53 "PDS factors (units in SL-1 words)" (page 159) for the memory store per item (PDS factor).	

Table 53 "PDS factors (units in SL-1 words)" (page 159) lists the memory store per item (PDS factor) used in calculating PDS requirements.

Table 53
PDS factors (units in SL-1 words)

Feature	Units
System overhead	32 768
500/2500-type telephones*	58
CLASS sets	58
M2006/2008 telephones*	105
M2216/2616 telephones*	115
M2317 telephones*	131
M3900 telephones	131
ACD telephones	16
IP Phones 200x	115
IP Softphones 2050	115
* See "Protected Memory for Phone Sets: Detail" (page 325).	

Feature	Units
Add-on modules	32
Templates	16
Consoles	236
DS/VMS Access TNs	14.5
ISDN BRI: • MISP cards • DSLs • TSPs • BRI DNs	 542 153 180 47
Analog trunks	54
Virtual Trunks	54
Trunk routes: • Constant term • Trunk routes	 1024 238
ISDN PRI/PRI2/ISL: • D-channels	 137
ISDN DTI/DTI2/JDMI: • DTI loops • DTI2 loops	 70 153
DISA DNs	18
Network: • Local loops • Remote loops	 91 95
ODAS	3
Customers: • Constant term • Customers	 1000 502
CPG Level Services	57
* See "Protected Memory for Phone Sets: Detail" (page 325).	

Feature	Units
Digitone receiver	12
Tone Detector	3
DN Translator (Consoles)	125
Author. Code (Custom.)	199
FGD ANI Database: • Constant term • NPA Codes	 43 547
CDP (Constant Term)	637
ACD/NACD: • ACD DNs • NACD DNs • ACD Positions	 92 174 src 115 dest 30
* See "Protected Memory for Phone Sets: Detail" (page 325).	

Unprotected data store

Table 54 "Unprotected data store" (page 161) describes the unprotected data store (UDS) area.

Table 54
Unprotected data store

Item	Calculation
Telephones (every type except BRI telephones)	Number of items **×** memory store per item where: memory store per item depends on the telephone type. For example: Number of 2500 telephones **×** memory store per 2500 telephone Number of telephones with display **×** memory store per display

Item	Calculation
BRI telephones	Constant term + (memory store per MISP × Number of MISPs) + (memory store per DSL × Number of DSLs) + (memory store per BRI line card × Number of BRI line cards) where: MISP = Multi-purpose ISDN Signaling Processor DSL = Digital Subscriber Loop
Analog trunks: • Paging trunks, RAN trunks, Add-on Data Module (ADM), RLA trunks, other analog trunks	Number of paging trunks × memory store per paging trunk Number of other analog trunks × memory store per other analog trunk and so on (Number of other analog trunks = Total number of analog trunks – Number of paging trunks – Number of RAN trunks – Number of ADMs – Number of RLAs)
Trunks (Call Detail Recording [CDR])	Total number of trunks × memory store per trunk
BRI trunks	Number of BRI trunks × memory store per BRI trunk
Trunk routes	(Number of trunk routes × memory store per trunk route) + (Total number of trunks ÷16) *Note:* Round up the division result.
DTI/DTI2/JDMI	Number of DTI loops × memory store per DTI loop Number of DTI2 loops × memory store per DTI2 loop
ISDN PRI/PRI2/ISL	

Item	Calculation
• PRI	(Number of D-channels × memory store per PRI D-channel) + (Number of output request buffers × memory store per output request buffer) + (2 × [Number of PRI trunks + Number of ISL trunks])
• PRI2	(Number of D-channels × memory store per PRI2 D-channel) + (Number of output request buffers × memory store per output request buffer) + (2 × [Number of PRI2 trunks + Number of ISL trunks])
I/O ports	(Number of TTYs × memory store per TTY) + (Number of CDR links × memory store per CDR link) + (Number of HS links × memory store per HS link) + (Number of APL links × memory store per APL link) + (Number of PMS links × memory store per PMS link) + (Number of Other links × memory store per Other link)
Other items (features):	Number of items × memory store per item
• Local loops, remote loops, secondary tapes, customer, TDS, MF sender, Conference card, DTR, Tone Detector, attendant, Peripheral Signaling card, LPIB, HPIB, background terminal, MSDL card	*Note:* The size of High Priority Input Buffer = Number of Groups × 32.
PBXOB and BCSOB	Number of Peripheral Signaling Cards × 640
DS/VMS access TNs	Memory store per DS/VMS TN × (Number of CallPilot ports + Number of data-only ports)
Application Module Link (AML)	Constant term + (Number of AMLs × memory store per AML)
Automatic Call Distribution (ACD):	
• Without ACD-C package	(Number of ACD DNs × 298) + (Number of ACD positions × 34)

Item	Calculation
• With ACD-C package	Additional memory size: (Number of ACD-C routes × 46) + (Number of ACD-C positions × 42) + [(Number of ACD-C DNs + Number of control directory numbers) × 80] + [(Number of ACD-C trunks + Number of ACD-C CRTs) × 30] + (Number of customers with ACD-C package × 240)
NARS/BARS/Coordinated Dialing Plan (CDP) **Assumption:** • If NTRF package is equipped, then Off Hook Queuing (OHQ) is also equipped	(Memory store per customer × Number of customers) + 2 × ([Number of route lists × memory store per route list] + [Number of routes with OHQ × memory store per route] + [Number of NCOS defined × memory store per NCOS])
Call registers **Assumptions:** • Call Register Traffic Factor = 1.865 • The formula for calculating the recommended number of call registers depends on traffic load for the system. • 28 CCS per ACD trunk	Call Registers memory size = Recommended number of call registers × memory store per call register 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 Loop Traffic • ISDN factor: $(1 - p)^2 + [4 \times (1 - p)] \times p + (3 \times p^2)$

Item	Calculation
If Total Voice Traffic > 3000 CCS, then:	Recommended number of call registers = (0.04 × Total Voice Traffic) + (0.18 × 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 sets) × 0.94] + {Number of ACD incoming trunks + [(Snacd + Tnacd) × 0.03]} × ISDN factor
IP Expansions	Additional memory is required as follows: • 2.0 MByte on the Main Cabinet/Chassis • 2.0 MByte on each survivable IP Expansion • 0.5 MByte on each non-survivable IP Expansion
System overhead	Memory store for system overhead

Table 55 "UDS factors (units in SL-1 words)" (page 165) lists the memory store per item (UDS factor) used in calculating UDS requirements.

Table 55
UDS factors (units in SL-1 words)

Feature	Units
System overhead	32 768
500/2500-type telephones	43.5
M2006/2008 telephones	89
M2216/2616 telephones	120
M2317 telephones	111.25
M3900 telephones	130
IP Phones 200x	120
IP Softphones 2050	96
Consoles	141
Add-on modules	24

Feature	Units
Displays	2
DS/VMS access TNs	16.5
ISDN BRI telephones: • Constant term • MISP cards • DSLs • BRI line cards	298 2270 264 96
Analog trunks: • RAN trunks Broadcast RAN trunks • RLA Trunks • AUTOVON Trunks • ADM • Other Analog Trunks	74 46 164 172 161
Virtual Trunks	161
Trunk routes	416
BRI trunks	148
Virtual Trunk D-Channel (DCIP)	850
DTI/DTI2 JDMI: • DTI loops • DTI2 loops	109 97
PRI/PRI2: • D-channels (PRI) • D-channels (PRI2)	836 850
I/O ports:	

Feature	Units
• I/O ports (total)	2085
• CDR links	
• HS links	128
• APL links	
• PMS links	143
• Other links	311
	130
	512
Local loops	69
Customers	243
Tone and Digit Switch	59
MF Sender	59
Conference cards	191
Digitone receiver	12
Tone Detector	13
Background terminals	96
MSDL cards	1395
AML (CSL):	
• Constant term	147
• AML Links	
	510
Call Registers	227

Mass storage

Small System software is stored in various areas of the NTDK20 Small System Controller (SSC) card. In terms of customer data, there are four possible areas where these records can be stored (see Figure 38 "Data storage on the NTDK20 SSC card" (page 168)):

- DRAM — stores and accesses the active version of customer records, system data, and overlay data

- primary flash drive C: — contains two copies of customer records (primary and backup records)

- backup flash drive Z: — retains the true backup copy of the customer database

- PC Card device A: or B: — if equipped, this 40 MByte device can store a complete backup copy of the customer database

Figure 38
Data storage on the NTDK20 SSC card
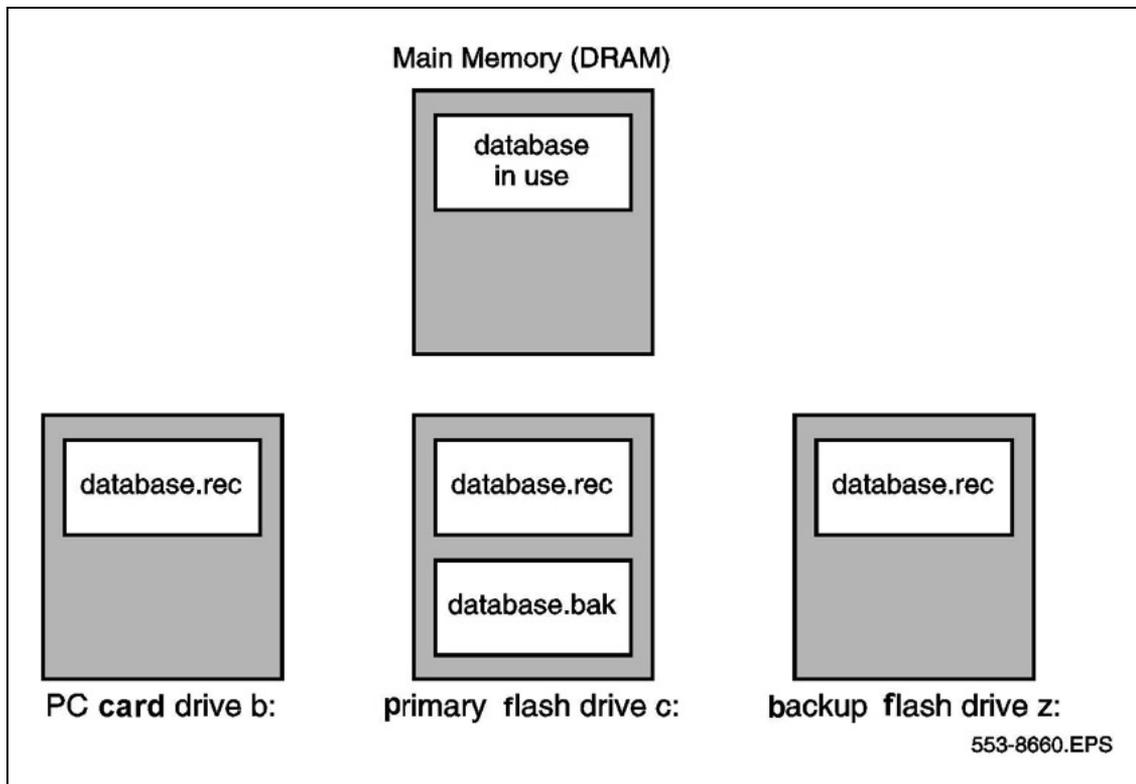
For the Small System, configuration data is both stored and loaded by accessing LD 43 and LD 143. The sequence of events where data is copied from one area to another depends on the status of the switch — new installation, software upgrade — and the purpose of the data transfer, such as to make a backup copy of the customer database.

Refer to *Communication Server 1000M and Meridian 1 Small System Maintenance (NN43011-700)* for information on database management, the Customer Configuration Backup and Restore (CCBR) feature, and LD 43 and LD 143 commands. For more information on the overlay commands, see also*Software Input Output Reference - Maintenance (NN43001-711)*.

Software installation

Software for new Small Systems is delivered on a preprogrammed Software Daughterboard or loads from a PC Card (Software Delivery card). Once this hardware is installed and the system is powered up, the system performs

a SYSLOAD and automatically invokes the Software Installation Program (LD 143). This menu-driven program assists in loading the software into the system.

Preprogrammed data

When a Small System is initially installed, customer data must be entered into the overlay programs. Telephones, for example, must be assigned features on their keys to allow them to function properly.

The SSC can be preprogrammed with customer data. If you load preprogrammed data into the system during installation, some overlay entries will be automatically configured on the telephones. For example, you can choose a telephone model that has predetermined feature and key assignments and a preassigned class of service. This can be a significant time-saver if there are numerous types of telephone models to program.

Preprogrammed data is not mandatory for software installation. In fact, the NTDK20 can be programmed with the minimum number of files to allow the Small System to operate.

Preprogrammed data cannot be removed from the Small System with one command once it is loaded into the system. However, items can be removed one by one after installation.

Preprogrammed data can be bypassed during first-time system installations. During start-up, the Software Installation Program is automatically invoked. The Small System loads system data from the NTDK20 SSC card and prompts the user for a variety of information, including the time and date, type of installation, feature set required, and type of database. At this point, if the user selects any response other than "Pre-Configured database," preprogrammed data is not loaded on the system.

> *Note:* The preprogrammed data on the system can provide an effective starting point for programming telephone and trunk information. Before bypassing the option of loading preprogrammed data, determine whether the default data can be used at this site.

Refer to Appendix "Preprogrammed data" (page 335) for a summary of the items preprogrammed in the Pre-Configured database on the Software Daughterboard. Refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)* for more detailed information about preprogrammed data and how to change it.

For information on upgrading to CS 1000 Release 4.5 software, refer to *Meridian 1 Small System Software-only Upgrade (NN43011-459)*.

Physical capacity

Resource constraints consist primarily of loop and card slot limitations. A fully expanded Cabinet system, with one Main and four Expansion Cabinets, provides 50 card slots for IPE cards. A fully expanded Chassis system, with one Main Chassis and four Expansion Chassis, each equipped with a Chassis Expander, provides 35 card slots for any IPE cards and 5 slots for NTDK16 48-port Digital Line Cards.

Each IPE circuit card has a loop entirely dedicated to it. Every group of four IPE card slots is programmed as an individual superloop.

There are a total of 640 timeslots (channels) for each Cabinet system. Each superloop provides 120 timeslots, while an IPE slot provides 30 timeslots.

In addition, there are five superloops available for use by either virtual or phantom TNs. This provides a maximum of 1248 TNs for the virtual/phantom TN space. There are an additional 1248 timeslots available for the virtual/phantom TNs.

For information about mapping loops, superloops, and card slots, see

Signaling and data links

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

Physical links

There are three types of physical links to consider:

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) recordsto a TTY or tape module. An SDI card has four ports.

Multi-purpose ISDN Signaling Processor (MISP)   The NTBK22 MISP card has four ports providing a combination of SDI, ESDI, and DCHI functions. Using MISP cards, the number of I/O ports in the system can reach 64. The data rate of each port of a MISP 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 at the NIC port is set at 100T (100 Mbps), 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 through 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.

For practical applications, the same data rate at the AML and ML is recommended.

Command and Status Link (CSL) The CSL is the version of AML specifically used for the communications between the system and the Meridian Mail (MM) system. It has some MM-specific messages. The interface is through an ESDI port. For Meridian Mail 1 through Meridian Mail 9, the CSL link rate was 4800 baud. Beginning with Meridian Mail 10, the link rate is 9600 baud.

OAM The system uses an SDI portto connect to a teletype (TTY) to receive maintenance commands or to print traffic reports, maintenance messages, or CDR records. CDR records can also be output directly to a magnetic tape system.

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 set 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 56 "I/O interface for applications" (page 172) 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 56
I/O interface for applications

Application	Type of link/ interface	Type of port	Sync or async
AML (associated set)	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 Mail	CSL	ESDI	Sync
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
Note: 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 set or trunk to its line card to a loop through the network to another loop, and on to another line or trunk card attachedto the terminating set or trunk.

This section discusses the following traffic considerations:

- "Loops and superloops" (page 173)

- "Lines and trunks" (page 174)

- "Service loops and circuits" (page 176)

- "Traffic capacity of Voice Gateway Media Cards" (page 178)

- "Traffic capacity engineering algorithms" (page 181)

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.

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" (page 181).

Lines and trunks

The relationship between lines and trunks is relevant for calculating loop requirements. Figure 39 "Traffic calls TDM only" (page 174) and Figure 40 "CS 1000M system call types" (page 175) show how traffic parcels on a loop can be broken up.

Figure 39 "Traffic calls TDM only" (page 174) represents traffic in a Meridian 1 TDM-based environment.

Figure 39
Traffic calls TDM only

553-3027
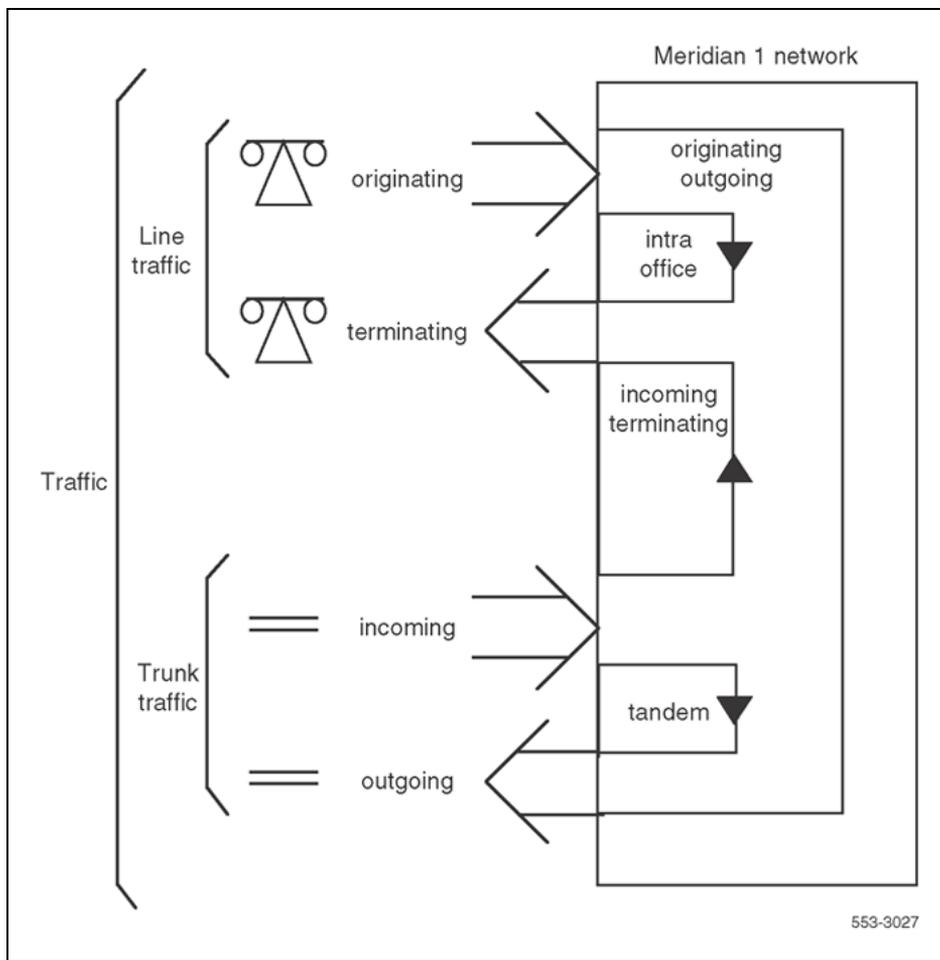
Voice over IP traffic

In the context of Voice over IP (VoIP) application, the lines include IP Phones and the trunks include IP Peer H.323 Virtual Trunks and Session Initiation Protocol (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 40 "CS 1000M system call types" (page 175) 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 connections.

Figure 40
CS 1000M system call types

553-AAA1643
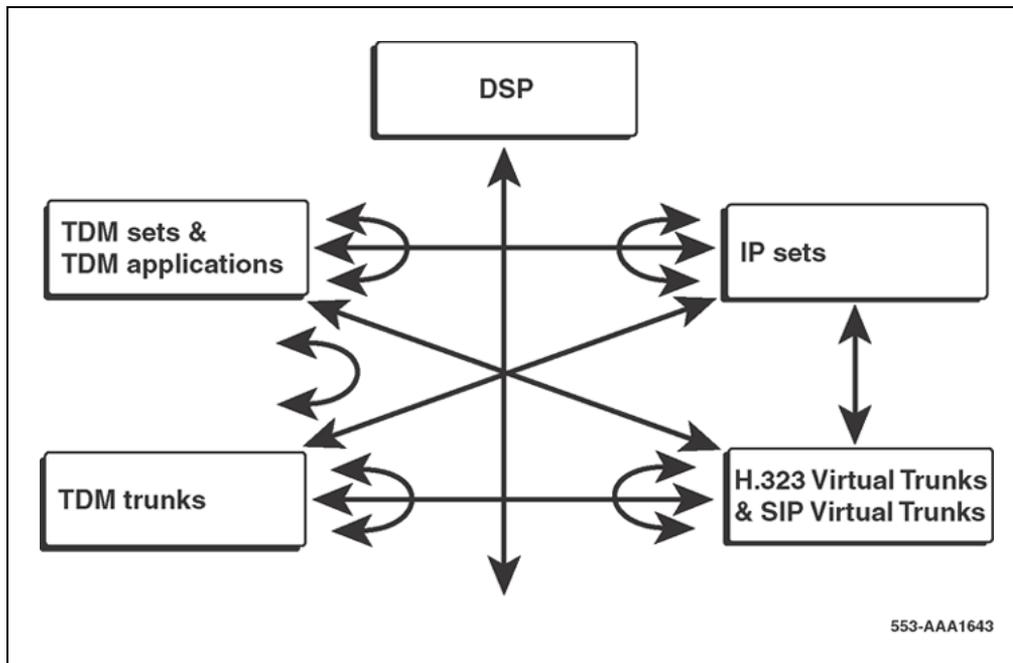
Table 57 "Connection type resources required" (page 175) lists the resources required for each type of connection.

Table 57
Connection type resources required

Connection Type	Resources
TDM to IP, IP to TDM	DSP
TDM to VT, VT to TDM	DSP and VT

Connection Type	Resources
IP to IP	no DSP
IP to VT or VT to IP	VT
TDM to TDM set or trunk calls	no DSP nor VT

Refer to "Resource calculations" (page 205) for the algorithms to calculate the required resources.

Service loops and circuits

Service circuits are required in call processing to provide specific functions to satisfy the requirements of a given application. 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" (page 176)
- "Conference" (page 176)
- "Broadcast circuits" (page 177)
- "DTR" (page 177)

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.

Conference

Each conference device on the SSC card provides 16 ports for conferencing. The SSC card provides 32 conference channels (ports). Each expansion daughterboard increases the total number of conference channels by 16. The maximum number of conference ports is 64.

A conference call can have three to six participants. Each conference participant requires one conference port. It is not possible to conference between conference devices.

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 16 simultaneous music users requires a CON loop, thus a Conference/TDS card, since these two service loops are not separable. For a small system, music users can share a conference loop with other applications. However, this is not a common practice in Call Center applications.

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 Appendix "Reference tables" (page 339)).

Broadcast circuits

The Nortel Integrated Recorded Announcer (Recorded Announcer) card provides either 8 or 16 ports to support Music, Recorded Announcement (RAN), and Automatic Wake Up.

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.

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 sets 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:

Step	Action
1	Calculate intraoffice DTR traffic:
	Intraoffice = 100 × DTR station traffic (CCS) ÷ AHT × (R ÷ 2) (Recall that R is the intraoffice ratio.)
2	Calculate outgoing DTR traffic:
	Outgoing = 100 × DTR station traffic (CCS) ÷ AHT × (1 - R ÷ 2)
3	Calculate direct inward dial (DID) DTR traffic:
	DID calls = DID DTR trunk traffic (CCS) × 100 ÷ AHT
4	Calculate total DTR traffic:
	Total = [(1.3 × 6.2 × intra) + (1.3 × 14.1 × outgoing calls) + (2.5 × DID calls)] ÷ 100
5	See "Digitone receiver load capacity 6 to 15-second holding time" (page 347) to determine the number of DTRs required. Note that a weighted average for holding times should be used.

—End—

Traffic capacity of Voice Gateway Media Cards

The number of IP Phones and IP Softphones is determined by the engineering of real-time usage, traffic capacity, network loop usage, and IPE slot usage.

Note 1: If a Media Card 32-port card or a Media Card 8-port card is running IP Line software, it is known as a Voice Gateway Media Card.

Note 2: The Voice Gateway Media Cards in a Small System support IP Phones type 2001, 2002, and 2004 and the IP Softphone 2050. In this section, the term "IP Phones" is used to refer generically to all these IP telephones.

Each Media Card 32-port card has 32 ports, which are used for establishing a voice connection between IP Phones and non-IP Phones. To configure a system as non-blocking (as is typically the case for ACD configurations), ensure that only 32 IP Phones are registered on each card.

Each Media Card 8-port card has 8 ports, which are used for establishing a voice connection between IP Phones and non-IP Phones. To configure a system as non-blocking (as is typically the case for ACD configurations), ensure that only 8 IP Phones are registered on each card.

A registered telephone is not synonymous with a configured telephone. When a telephone is registered, it is as if the telephone is plugged in. When the telephone de-registers, it is as though the telephone is unplugged. Registration consists of two steps:

Step	Action
1	Verifying that the user's TN is valid and has not yet been registered.
2	Associating the TN on the system.

—End—

If an IP Phone is unplugged, it is automatically unregistered after a predetermined time-out.

Voice Gateway Media Cards in a system are pooled by customer number, are assigned dynamically, and are allocated preferentially by matching bandwidth management zones.

Note 1: The average number of Busy Hour Call Attempts (BHCA) must not exceed 1200 per Voice Gateway Media Card.

Note 2: The capacity of a 32-port Media Card at P.01 GoS = 794 CCS.

Refer to the following three examples for further clarification.

Example 1   One hundred fifty (150) IP Phones with "typical" business usage of 600 call seconds per hour (6 CCS) for each telephone on average (for example, 5 calls of 120 seconds duration per hour):

- 150 × 6 CCS = 900 CCS

- Two Media Card 32-port cards are required.

  Two Media Card 32-port cards support up to 1738 CCS.

Example 2 Five hundred (500) IP Phones with "heavy" business usage of 12 CCS for each telephone on average (for example, 6-7 calls of 180 seconds duration every hour):

- 500 × 12 CCS = 6000 CCS

- Six Media Card 32-port cards are required.

  Six Media Card 32-port cards support up to 6013 CCS

Example 3 Forty-eight (48) Call Center Agents with an allocation of 36 CCS for each telephone:

- 48 ports required ÷ 32 ports for each Media Card 32-port card = 2 Media Card 32-port cards (1.5 must be rounded up to 2)

- Two Media Card 32-port cards are required.

  Note: For Call Center Agents, Nortel recommends that one port be provisioned for each agent.

Gateway channels traffic engineering

Configure no more than four Voice Gateway Media Cards on each superloop to eliminate the possibility of blocking because of insufficient timeslots (for example, 4 Voice Gateway Media Cards × 32 ports = 128 timeslots). Use Table 58 "Voice Gateway Media Card recommendations based on CCS capacity" (page 180) to determine the number of Voice Gateway Media Cards required to maintain the recommended capacity.

Table 58
Voice Gateway Media Card recommendations based on CCS capacity

Number of cards	Media Card 8-port card CCS capacity	Media Card 32-port card CCS capacity
1	113	794
2	319	1822
3	550	2891

Note 1: CCS is the number of hundred call seconds per hour.

Note 2: The IP Phone blocking probability is P.01.

Note 3: If the number of Media Card 32-port cards exceeds 6, or the number of Media Card 8-port cards exceeds 24, use the following formula to estimate capacity: (CCS .÷ 6192) x 192

Number of cards	Media Card 8-port card CCS capacity	Media Card 32-port card CCS capacity
4	794	3982
5	1044	5083
6	1300	6192
7	1559	(see Note 3)
8	1822	(see Note 3)
9	2088	(see Note 3)
10	2354	(see Note 3)
12	2891	(see Note 3)
16	3982	(see Note 3)
20	5083	(see Note 3)
24	6192 (see Note 3)	(see Note 3)

Note 1: CCS is the number of hundred call seconds per hour.

Note 2: The IP Phone blocking probability is P.01.

Note 3: If the number of Media Card 32-port cards exceeds 6, or the number of Media Card 8-port cards exceeds 24, use the following formula to estimate capacity: (CCS .÷ 6192) x 192

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. Post-dialing delay is less than 1.5 seconds on all calls.

Traffic models

Table 59 "Traffic models" (page 182) summarizes the traffic models that are used in various subsystem engineering procedures.

Table 59
Traffic models

Model	Assumptions	Service criteria	Applicability
Erlang B	Infinite sources (ratio of traffic sources to circuits > 5:1)	Blocked calls cleared (no queueing)	Loop, ringing circuit blocking
Erlang C	Infinite sources	Blocked calls delayed Infinite queue	Dial tone delay, I/O buffers, Digitone, RAN trunks
Poisson	Infinite sources	Blocked calls held for a fixed length	Incoming/outgoing trunks, Digitone, Call Registers, RAN trunks

Typically, the GoS for line-side traffic is based on Erlang B (or Erlang Loss formula) at P.01 GoS. When there is no resource available to process a call entering the system, the call is blocked out of the system. Therefore, the correct model to calculate the call's blocking probability is a "blocked call cleared" model, which is the basis of Erlang B.

When a call is already in the system and seeking a resource (trunk) to go out, the usual model to estimate trunk requirements is based on the Poisson formula. The reasons are:

- The Poisson model is more conservative than Erlang B (in that it projects a higher number of circuits to meet the same GoS). This reflects trunking requirements more accurately, since alternative routing (or

routing tables) for outgoing trunk processing tends to increase loading on the trunk group.

- General telephony practice is to provide a better GoS for calls already using system resources (such as tones, digit dialing, and timeslots). Incomplete calls inefficiently waste partial resources. With more trunk circuits equipped, the probability of incomplete calls is lower.

Real-time capacity

Real-time capacity (load) refers to the ability of the Call Server to process instructions resulting from calls in accordance with service criteria.

Existing systems can use methods based on traffic data in order to determine Rated Call Capacity and current utilization levels. Refer to *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for a description of the TFS004 call capacity report and for information on interpreting TFS004 output.

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

Equivalent Basic Calls

An Equivalent Basic Call (EBC) is a measure of the real time required to process a basic call. A basic call is defined as a simple, unfeatured call between two 2500-type sets on the same switch using a four-digit dialing plan. The terminating set is allowed to ring three times, then is answered, waits approximately two seconds, and hangs up. The originating set 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 60 "Real-time capacity (EBC) of Small System CPU, by software release" (page 184) gives the rated capacity of the SSC processor.

Table 60
Real-time capacity (EBC) of Small System CPU, by software release

	SSC (EBC)	
Release	**No IP expansion**	**With IP expansion**
25.40	42 000	n/a
Succession 3.0 Software	42 000	35 000
CS 1000 Release 4.5	42 000	35 000

Network delay

There is some impact on real-time performance (estimated to be 20%) when digital trunks are installed in IP expansion cabinets or chassis. However, there is still sufficient real time to support five fully configured cabinets in a typical business configuration.

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 Nortel Enterprise Configurator (NNEC) 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 set with incoming CDR is modeled as a basic call plus a real-time increment for incoming DID plus an increment for digital sets 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 total of the impacts of all features, plus the number of calls, is the real-time load on the system, in EBC.

For penetration and real-time factors and for the detailed EBC calculations, refer to "System calls" (page 210) and "Real-time calculations" (page 212).

Call Server real-time calculations

The system EBC divided by the processor's rated capacity (see Table 60 "Real-time capacity (EBC) of Small System CPU, by software release" (page 184)) 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 TMDI 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" (page 249).

Real-time algorithm

As described above, calculating the real-time usage of a configuration requires information on the number of busy hour call attempts and the penetration factors of each feature.

Busy hour calls

If the switch is already running, the number of busy hour calls or call load can be determined from the traffic printout TFS004. The second field of this report (after the header) contains a peg count of CP Attempts. Examine a period of several days (a full week, if possible) to determine the maximum number of CP attempts experienced. This number varies with season, as well. The relevant number is the average of the highest ten values from the busiest four-week period of the year. An estimate is acceptable, 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 sets and trunks must be made. Refer to Table 64 "Major parameters for VoIP resource calculations" (page 206) for a description of the parameters that are required and default values, if applicable.

Telephones

As the primary traffic source to the system, telephones have a unique real-time impact on the system. For the major types in the following list, 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 sets

- Digital: M2000 series Meridian Modular Telephone, voice and/or data ports

- Consoles

- IP Phone 2001, IP Phone 2002, IP Phone 2004, IP Phone 2007, IP Audio Conference Phone 2033

- IP Softphone 2050

Trunks

Depending on the type of trunk and application involved, trunks can either be traffic sources, which generate calls to the system, or resources that satisfy traffic demands. Default trunk CCS in an office environment is 26 CCS. Call Center applications 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 AHT_{RAN} is 30 seconds.

Music
The default value for AHT_{MUSIC} is 60 seconds.

Signaling Server

The following software components operate on the Signaling Server:

- Terminal Proxy Server (TPS)

- H.323 Gateway (Virtual Trunk)

- SIP Gateway (Virtual Trunk)

- Network Routing Service (NRS)

- CS 1000 Element Manager web server

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

A Signaling Server can also function as an application server for the Personal Directory, Callers List, and Redial List applications and Password administration. See "Application server for Personal Directory, Callers List, and Redial List" (page 192).

Table 61 "Elements in Signaling Server" (page 188) describes the function and engineering requirements of each element. For detailed Signaling Server engineering rules and guidelines see "Signaling Server algorithm" (page 226).

Table 61
Elements in Signaling Server

Element	Function and engineering requirements
Terminal Proxy Server (TPS)	• The TPS handles initial signaling exchanges between an IP Phone and the Signaling Server. • The TPS supports a maximum of 5000 IP Phones on each Signaling Server. • The TPS manages the firmware for the IP Phones that are registered to it. Accordingly, the TPS also manages the updating of the firmware for those IP Phones. • The redundancy of TPS is N+1. Therefore, one extra Signaling Server can be provided to cover TPS functions from N other servers.
Note: The feasibility of combining the Terminal Proxy Server, H.323 Gateway, SIP Gateway, and Network Routing Service on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

Element	Function and engineering requirements
H.323 Gateway (Virtual Trunk)	• The IP Peer H.323 Gateway trunk, or H.323 Virtual Trunk, provides the function of a trunk route without a physical presence in the hardware. The H.323 Gateway supports direct, end-to-end voice paths using Virtual Trunks. • The H.323 Signaling software (Virtual Trunk) provides the industry-standard H.323 signaling interface to H.323 Gateways. It supports both en bloc and overlap signaling. This software uses an H.323 Gatekeeper to resolve addressing for systems at different sites. • The H.323 Gateway supports up to 1200 H.323 Virtual Trunks per Signaling Server, assuming a combination of incoming and outgoing H.323 calls (see "Maximum number of SIP and H.323 Virtual Trunks" (page 191)). Beyond that, a second Signaling Server is required. *Note 1:* At least 512 MByte of memory is required on the Signaling Server to obtain 1200 H.323 Virtual Trunks. If the Signaling Server has less than 512 MByte of memory, then a maximum of 382 Virtual Trunks can be configured. *Note 2:* If H.245 tunneling is not enabled, then a maximum of 900 H.323 Virtual Trunks can be supported on a Signaling Server equipped with at least 512 MByte of memory. • The redundancy mode of the H.323 Gateway is 2 × N. Two H.323 Gateways handling the same route can provide redundancy for each other, but not for other routes.
SIP Gateway (Virtual Trunk)	• The SIP Gateway trunk, or SIP Virtual Trunk, provides a direct media path between users in the CS 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 1800 SIP Virtual Trunks (see "Maximum number of SIP and H.323 Virtual Trunks" (page 191)). • 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.
Note: The feasibility of combining the Terminal Proxy Server, H.323 Gateway, SIP Gateway, and Network Routing Service on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

Element	Function and engineering requirements

Network Routing Service (NRS)	• The NRS has three components:
	  — H.323 Gatekeeper
	  — SIP Redirect Server
	  — Network Connection Service (NCS)
	
	• The NRS must reside on the Leader Signaling Server. In a redundant configuration, the NRS is configured as Primary, Alternate, or Failsafe (if required).
	• The NRS software limit for the combined total number of endpoints and routing entries is 20 000. The limit for the total number of endpoints is 5000 (up to 5000 SIP and up to 2000 H.323 endpoints).
	• The redundancy of the NRS is in a mode of 2 × N. An alternate NRS can serve only the NRS it is duplicating.

• H.323 Gatekeeper	• All systems in the network register to the H.323 Gatekeeper, which provides telephone number to IP address resolution. • The capacity of the H.323 Gatekeeper is limited by the endpoints it serves and the number of entries at each endpoint. • Potential hardware limits are the Signaling Server processing power and memory limits. • Since the Gatekeeper is a network resource, its capacity is a function of the network configuration and network traffic (IP calls). Some basic network information is required to engineer a Gatekeeper.
• SIP Redirect Server	• The SIP Redirect Server provides telephone number-to-IP address resolution. It uses a Gateway Location Service to match a fully qualified telephone number with a range of Directory Numbers (DN) and uses a SIP gateway to access that range of DNs.
• Network Connection Service (NCS)	• The NCS provides an interface to the TPS, enabling the TPS to query the NRS using the UNIStim protocol. The NCS is required to support the Media Gateway 1000B, Virtual Office, and Geographic Redundancy features.
Element Manager	• Has a negligible impact on capacity and can reside with any other element.
Note: The feasibility of combining the Terminal Proxy Server, H.323 Gateway, SIP Gateway, and Network Routing Service on a Signaling Server is determined by traffic associated with each element and the required redundancy of each function.	

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 FDs for Virtual Trunks is 1800.

- Each SIP call uses one FD.

- Each incoming H.323 call uses two FD.

- Each outgoing H.323 call uses one FD.

When no more FDs are available (available FD = 0), new channels added on the Call Server will not be able to register on the Signaling Server.

Each Signaling Server supports up to 1800 Virtual Trunks. The maximum number of SIP and H.323 trunks will depend on traffic patterns, both the split between SIP and H.323 calls and the split between incoming and outgoing H.323 calls. gives examples of the maximum number of Virtual Trunks supported for different configurations.

Table 62
Maximum number of Virtual Trunks, per Signaling Server

	H.323*			Total Virtual Trunks
SIP	Incoming	Outgoing	Total H.323	
1800	0	0	0	1800
0	600	600	1200	1200
0	900	0	900	900
600	0	1200	1200	1800
600	300	600	900	1500
*Assumes H.245 tunneling enabled.				

The formula to calculate the maximum number of Virtual Trunks is:

(Num_of_SIP × 1 FD) + (Num_of_Incoming_H323 × 2 FD) + (Num_of_Outgoing_H323 × 1 FD) <= Max_Num_of_FDs
where Max_Num_of_FDs = 1800

Impact of H.245 tunneling By default, H.245 tunneling is on. Unless there is a specific reason to disable tunneling, such as for maintenance, it should always be on. When tunneling is off, the handling capacity of the Signaling Server is reduced to a maximum of 900 H.323 Virtual Trunks.

Application server for Personal Directory, Callers List, and Redial List

The database for the Personal Directory, Callers List, and Redial List features for IP Phones must be located on one Signaling Server. The applications cannot be divided: all users in a system will either have the combined Personal Directory, Callers List, and Redial List features or no feature at all. The Signaling Server can support a database for up to 15 000 users.

The storage capacities for the Personal Directory, Callers List, and Redial List features for each user are as follows:

- **Personal Directory:** Stores up to 100 entries per user of user names and DNs.

- **Callers List:** Stores up to 100 entries per user of caller ID information and most recent call time.

- *Redial List:* Stores up to 20 entries per user of dialed DNs and received Call Party Name Display with time and date.

The Signaling Server requires a minimum of 512 MByte of memory in order to support the Personal Directory, Callers List, and Redial List applications.

If the system size is relatively small, in terms of number of users as well as calling rates, one Signaling Server can serve both database and normal Signaling Server functions. With the Personal Directory, Callers List, and Redial List database co-resident with other applications (TPS, H.323/SIP Gateways, NRS, Element Manager), a Signaling Server with 512 MByte of memory can serve up to1000 IP users and 382 Virtual Trunks.

There is no redundancy for the Signaling Server dedicated to the Personal Directory, Callers List, and Redial List database. If that Signaling Server fails, the system will lose those applications. However, the other Signaling Servers will continue to function normally without the Personal Directory, Callers List, and Redial List features.

Software configuration capacities

The tables in "Design parameters" (page 143) 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.

CS 1000M capacities

Since IP telephony consumes more processing than TDM, the total number of sets 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 63 "CS 1000M Small System traffic capacities summary" (page 193)
summarizes the capacities of CS 1000M Small Systems. Values in each
cell indicate the total number of sets 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.

Note: Values in each cell are exclusive, not additive.

Table 63
CS 1000M Small System traffic capacities summary

Call server	Platform name	Total number of sets				
		Pure TDM (no trunking)	IP sets with access to PSTN	Pure IP (no access to PSTN)	Mixed TDM and IP sets	Max VTNs
MSC (see Notes 1 and 2)	CS 1000M Chassis (Mini)	160	800	800	144 TDM 300 IP	1248
SSC (see Note 2)	CS 1000M Chassis	720	1000	1000	640 TDM 800 IP	1248
SSC (see Note 2)	CS 1000M Cabinet	720	1000	1000	600 TDM 1000 IP	1248
SSC (see Note 2)	CS 1000S	480	1000	1000	400 TDM 700 IP	1248

Note 1: With MSC, systems are limited to a maximum of 2000 Corporate Directory entries.

Note 2: With MSC or SSC with 16 MByte DRAM, systems approaching or exceeding 1000 total TNs (including Virtual TNs) should monitor Available Memory and, if memory falls below 32 000 words, should upgrade to a suitable NTDK20 SSC card.

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 1000S and CS 1000M systems with IP Peer H.323 Gateways, IP Line 3.0, IP Trunk 3.01 (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 41
Bandwidth management example
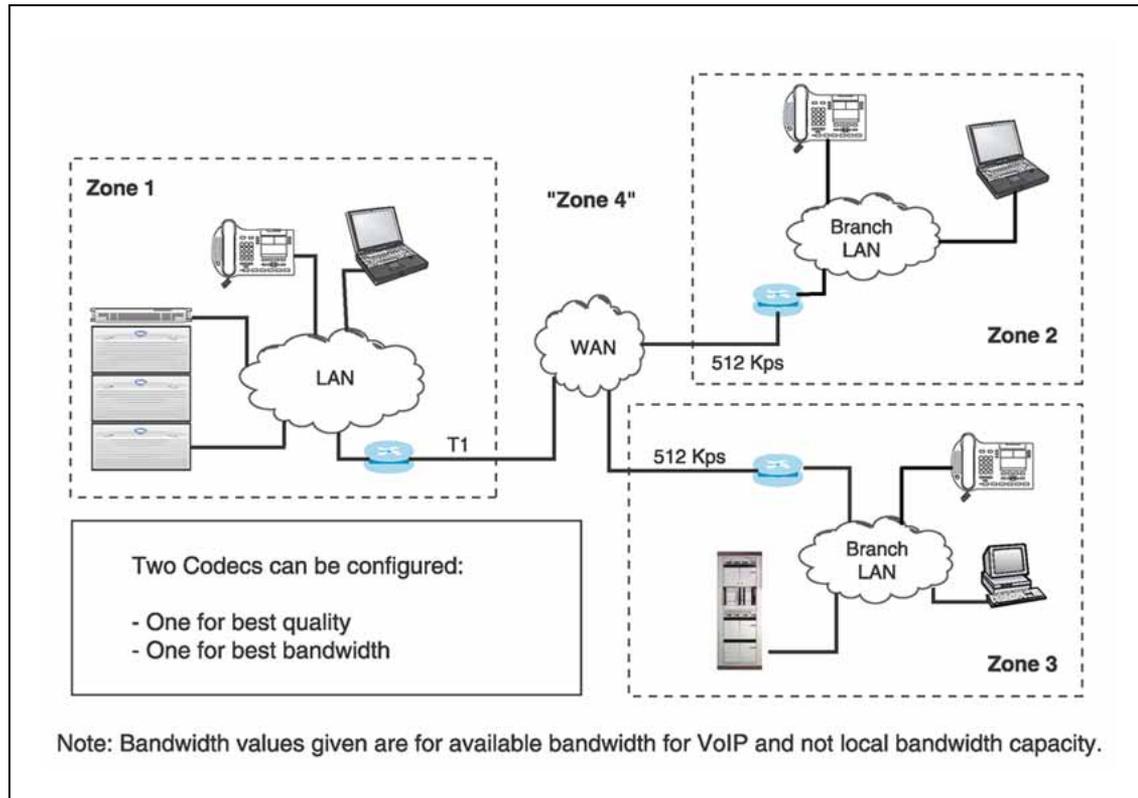
CS 1000 provides support for bandwidth management on a network-wide basis so that voice quality can be managed between multiple Call Servers using IP Peer Networking in certain scenarios.

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 in which 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 in which 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 MG 1000B IP Phones 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 MG 1000B SSC.

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 Voice Gateway 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 Voice Gateway Media Cards. 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 or Leader Voice Gateway Media Card. All the other Signaling Servers, Voice Gateway Media Cards are defined as Followers.

A node does not by necessity need a Signaling Server or Media Card. A node can be either all Signaling Servers (a practical maximum of 5, with an actual maximum of 35), or all Media Cards (a maximum of 30), or both (a maximum of a total of 35 elements).

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.

It is possible to have TPS configured Signaling Servers running in more than one node. In this case, the sets are configured to the node to which they must register.

Voice Gateway Media Card is a term used to encompass the Media Card 32-port line card, Media Card 8-port line card, and ITG-P 24-port line card. These cards plug into an Intelligent Peripheral Equipment (IPE) shelf in the Meridian 1 and CS 1000M systems, and into a Media Gateway 1000S and Media Gateway 1000S Expander in the CS 1000S system. They also plug into MG 1000E for the CS 1000E systems.

In IP Line 4.5, the LTPS executes on the Signaling Server and the voice gateway executes on the Voice Gateway Media Cards. All IP Phones register with the Signaling Server. The Voice Gateway Media Cards only provide access to the voice gateway. 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 382, the H.323 Gateway cannot co-reside with any application. In this case, a standalone Signaling Server is required to run the H.323 Gateway application on its Leader Signaling Server in a new node. Different customers require different H.323 Gateways (separate nodes).

A single Signaling Server can support multiple routes, with all the routes on a single Signaling Server (on a node) configured to use same node ID and same D channel.

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 382, the SIP Gateway cannot co-reside with any application. In this case, a standalone Signaling Server is required to run SIP Gateway application on its Leader Signaling Server in a new node.

Different customers require different SIP Gateways (separate nodes). A single Signaling Server can support multiple routes, with all the routes on a single Signaling Server (on a node) configured to use same node ID and same D channel.

From a Signaling Server configuration point of view, PD/RL/CL can reside on Leader or Follower Signaling Server. There is no relationship between PD and nodes, and it is possible to run PD on any Signaling Server. When PD is standalone, it is recommended that the PD be placed in its own node as the Leader to simplify the organization of devices and their presentation in EM.

It is also recommended to run NRS on the Leader Signaling Server. There is no limitation in the NRS software to run NRS application on the Signaling Server Follower. However, since the installation of the Signaling Server Follower does not allow the NRS configuration, the NRS must be configured after the installation procedure.

In case of Signaling Server applications running co-res, when the number of IP Phones is greater than 1000, and the number of virtual trunks is greater than 382, a separate Signaling Server is required to run the NRS application on its Leader Signaling Server.

Since a single NRS is required for the whole network, the recommendation is to configure NRS on a separate node. The alternate recommendation for NRS is to run on the Signaling Server Leader, and configure on a separate customer site.

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.

One Signaling Server is a Leader Signaling Server that acts as the primary, or master, TPS. The other Signaling Server is a Follower Signaling Server that acts as a secondary, redundant TPS. The NRS, H.323/SIP Gateways must reside on the Leader Signaling Server.

The redundancy of TPS is N+M. Therefore, extra Signaling Server(s) can be provided to cover TPS functions from N other servers. With a redundant Load Sharing Signaling Server:

- One or more Signaling Server(s) can be configured in a normal configuration.

- The redundant Signaling Server(s) must be configured in the same TPS node as the Signaling Servers they are protecting.

- If any of the Signaling Servers fails, sets that were registered to the failed Signaling server register to the remaining Signaling servers in the same node.

- If all Signaling servers in a node running TPS fail and the node has Voice Media Gateway Cards configured, one of the Voice Gateway Media Cards is elected to be the node Master and the other cards will be followers. IP sets will register to these Voice Media Gateway Cards up to 128 sets per card.

In a case where no backup Signaling Server exists:

- If there is no backup Signaling Server, and the primary Signaling Server fails, one of the Voice Gateway Media Cards is elected to be the node Master.

- The IP Phones then register to the Voice Gateway Media Cards.

The redundancy of VGMCs is N+M. The limit of 30 VGMCs per node does not impact redundancy. If the DSPs on the VGMC 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 Gateway is either 1:1 or 1+ M. In the 1:1 configuration one operational Gateway, and one in idle state, ready to take over. Two H.323 Gateways handling the same route can provide redundancy for each other, but not for other routes.

In the 1 + M configuration, M additional Signaling Servers are provided as leaders in their own node, and each of these 1 + M Signaling Servers support trunks routes that are in the same route list. To balance the load they have the same cost factor.

The 1 + M configuration provides performance improvement over the 1:1 configuration because fewer resources are lost in a single failure and the redundant routes are immediately available to carry traffic. The 1+ M configuration does bear the cost of additional virtual trunk licenses.

The redundancy of H.323 Gateway is provided within a node. The Primary Gateway runs on the (redundant) Signaling Server Leader on the Signaling Server Follower. The same rules apply for SIP as for H.323 Gateway.

There is a single PD/RL/CL application that must be located on a single Signaling Server. The applications cannot be divided: all IP clients in a system will either have the combined Personal Directory, Callers List, and Redial List features, or no feature at all. There is no active redundant PD/RL/CL application. A backup and restore function is provided to preserve customer data and load it into a new Signaling Server. Failure of PD/RL/CL Signaling server does not impact call processing.

In a redundant configuration, the NRS is configured as Primary, Alternate, or Failsafe. Although a network requires only one (Primary) NRS, Nortel recommends that an Alternate NRS, and Failsafe NRS, be configured in the network. The Failsafe NRS is, by default, running on SIP/H.323 Gateway.

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 VGMC cards, a separate node must be configured.

A Signaling Server with Co-Res application can support up to 1000 sets and 382 virtual trunks and Virtual Trunks and NRS applications must run on a Leader Signaling Server. 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 Voice Media Gateway 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 3.01 (and later) 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 3.01 (and later) 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 3.01 (and later) 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 3.01 (and later) 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 3.01 (and later) node A1 while all other 408-xxx-xxxx calls are routed to IP Trunk 3.01 (and later) 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 VGMCs and Signaling Servers combined in a node is limited to 35, without exceeding the limitations of each element type. The maximum number of VGMCs per node is 30 while the maximum number of Signaling Servers per 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, CS 1000S

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 on Zone/IP Telephony node engineering, refer to the following NTPs:

-   *Converging the Data Network with VoIP Fundamentals (NN43001-260)*

-   *Branch Office Installation and Commissioning (NN43001-314)*

-   *IP Trunk Fundamentals (NN43001-563)*

-   *IP Line Fundamentals (NN43001-500)*

Resource calculations

Contents

This section contains information on the following topics:

Introduction

This chapter describes the algorithms implemented by the Nortel Enterprise Configurator (NNEC) tool in order to calculate the resources required by the system.

In many cases, the calculations require user inputs that are the result of pre-engineering performed in accordance with the capacities and guidelines described in "System capacities" (page 153) and "Application engineering" (page 249).

Note: 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 64 "Major parameters for VoIP resource calculations" (page 206) describes the major parameters used in the Voice over IP (VoIP) calculations. Some are user input and others are calculated.

Table 64
Major parameters for VoIP resource calculations

Parameter	Description	Equation	Default value
TDM set CCS (L_{TDM})	Sum of all digital and analog set and line-side T1/E1 ports, in CCS	(Number of digital sets + Number of analog sets + Number of line-side T1/E1 ports) × CCS per set	CCS per set: 5
IP set CCS (L_{IP}) (See Note 3 at end of table.)	Sum of all IP and IP ACD agent sets, in CCS	[(Number of IP sets – Number of IP ACD agents) × CCS per IP set] + (Number of IP agent sets × CCS per set	CCS per set: 5
MDECT set CCS (L_{DECT})	Sum of all MDECT mobile sets, in CCS	Number of MDECT sets x CCS per set	CCC per set: 5
Total line CCS (L_{CCS})	Sum of all TDM, IP, and MDECT set CCS	TDM set CCS (L_{TDM}) + IP set CCS (L_{IP}) + MDECT set CCS (L_{DECT})	
TDM trunk CCS (T_{TDM})	Sum of all analog and digital trunks, in CCS	(Number of analog trunks + Number of digital trunks) × CCS per trunk	CCS per set : 26
Converged Desktop ratio (r_{DTP})	Of the total number of sets, the portion that have the Converged Desktop feature	(Number of sets with Converged Desktop) ÷ (Total number of sets)	
Converged Desktop CCS (V_{DCCS}) (See Note 2 after the table.)	Converged Desktop CCS calculated as a percentage of total line CCS	Total line CCS (L_{CCS}) × Converged Desktop ratio (r_{DTP})	
SIP CTI TR/87 ratio rMO	Of the total number of sets, the portion that have the SIP CTI/TR87 feature	Number of sets with SIP CTI/TR87 feature ÷ Total number of sets	
H.323 Virtual Trunk CCS (HVT_{CCS})	Sum of all H.323 Virtual Trunks, in CCS	Number of H.323 Virtual Trunks (VT_{323}) × CCS per VT_{323}	

Parameter	Description	Equation	Default value
SIP Virtual Trunk CCS (SVT_{CCS})	Sum of all SIP Virtual Trunks, in CCS	Number of SIP Virtual Trunks (VT_{SIP}) × CCS per VT_{SIP}	
H.323 Virtual Trunk ratio (v_H)	Of total Virtual Trunk CCS, the portion that are H.323 Virtual Trunks	H.323 Virtual Trunk CCS (HVT_{CCS}) ÷ [H.323 Virtual Trunk CCS (HVT_{CCS}) + SIP Virtual Trunk CCS (SVT_{CCS})]	
SIP Virtual Trunk ratio (v_S)	Of total Virtual Trunk CCS, the portion that are SIP Virtual Trunks	SIP Virtual Trunk CCS (SVT_{CCS}) ÷ [H.323 Virtual Trunk CCS (HVT_{CCS}) + SIP Virtual Trunk CCS (SVT_{CCS})]	
Virtual Trunk CCS (VT_{CCS})	Sum of H.323 Virtual Trunk CCS and SIP Virtual Trunk CCS	H.323 Virtual Trunk CCS (HVT_{CCS}) + SIP Virtual Trunk CCS (SVT_{CCS})	
Total trunk CCS (T_{TCCS})	Sum of all Virtual Trunk CCS and TDM trunk CCS	Virtual Trunk CCS (VT_{CCS}) + TDM trunk CCS (T_{TDM})	
Local CallPilot CCS (CP1) (See Note 4 after the table.)	CallPilot calls within the local node, calculated from number of local CallPilot ports	Local CallPilot ports × CCS per port ($CP1_{CCS}$)	
Network CallPilot CCS (CP2) (See Note 4 after the table.)	Network CallPilot calls to the local node, calculated from number of network CallPilot ports	Network CallPilot ports × CCS per port ($CP2_{CCS}$)	
IP ratio (P) (See Note 3 after the table.)	Of total line CCS, the portion that are from IP sets	IP set CCS (L_{IP}) ÷ Total line CCS (L_{CCS})	
Virtual Trunk ratio (V) (See Note 3 after the table.)	Of total trunk CCS, the portion that are from Virtual Trunk access ports	Virtual Trunk CCS (VT_{CCS}) ÷ Total trunk CCS (T_{TCCS})	
Total system CCS (T_{CCS})	Sum of all line and trunk CCS	Total line CCS (L_{CCS}) + Total trunk CCS (T_{TCCS})	
Intraoffice ratio (R_I)	Of the total number of calls, the portion that are set-to-set calls		0.30

Parameter	Description	Equation	Default value
Tandem ratio (R_T) (See Note 5 after the table.)	Of the total number of calls, the portion that are trunk-to-trunk calls		0.05
Incoming ratio (I)	Of the total number of calls, the portion that are trunk-to-set calls		0.40
Outgoing ratio (O)	Of the total number of calls, the portion that are set-to-trunk calls		
Average holding time (AHT_{XX})	Average holding time for different call types: Set-to-set (AHT_{SS}) Trunk-to-set (AHT_{TS}) — also used for ACD agents (AHT_{AGT}) Set-to-trunk (AHT_{ST}) Trunk-to-trunk (AHT_{TT})		60 sec 150 sec 150 sec 180 sec
Weighted average holding time (WAHT)		$(R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$	
Total calls (T_{CALL})	Total system calls per hour	$0.5 \times T_{CCS} \times 100 \div WAHT$	
Intraoffice calls (C_{SS})	Number of set-to-set calls	$R_I \times T_{CALL}$	

Parameter	Description	Equation	Default value
Tandem calls (C_{TT})	Number of trunk-to-trunk calls	$R_T \times T_{CALL}$	
Originating/outgoing calls (C_{ST})	Number of set-to-trunk calls	$O \times T_{CALL}$	
Terminating/incoming calls (C_{TS})	Number of trunk-to-set calls	$I \times T_{CALL}$	
DSP calls (C_{DSP})	Number of calls involving DSP		
Virtual Trunk calls (C_{VT})	Number of calls involving Virtual Trunks		
Conference loop ratio (r_{Con})	Ratio of conference loops to traffic loops	(Number of conference loops) ÷ (Total number of loops)	0.07

Note 1: In order to use the system traffic equations, all line-side T1/E1 and PRI trunks must be converted to number of ports. To convert T1 to ports: number of cards × 24. To convert E1 to ports: number of cards × 30.

Note 2: Converged Desktop traffic is part of the SIP Virtual Trunk traffic. The parameter value V_{DCCS} must be less than the capacity of the number of SIP ports (VT_{SIP}).

Note 3: A site is considered to be a call center when the proportion of ACD agent sets exceeds 15% of the total sets in the system. For call centers, ACD agent calls are included in the calculations for Call Server usage. However, they are initially excluded from the calculations for DSP and Virtual Trunk resources. After the DSP and Virtual Trunk resources have been calculated for non-ACD (reduced) traffic, the resources required to support the non-blocking ACD application (one DSP port for each ACD agent) are added back in to the results, in order to obtain the total system DSP and Virtual Trunk requirements. The IP ratio (P) is modified for the non-ACD part of the calculation: P' = (L_{IP} without ACD)/(L_{TDM} without ACD + L_{IP} without ACD + L_{DECT}). The Virtual Trunk ratio (V) remains unchanged. The default traffic value for ACD agent sets (TDM and IP) is 33 CCS per set.

Note 4: CallPilot message traffic is embedded in total line traffic. To calculate the real-time impact on the Call Server, CallPilot ports are converted to calls. Only CallPilot ports serving the local node (CP1) and handling network traffic (CP2) have a real-time impact on the Call Server.

Note 5: The tandem ratio should be kept at a relatively small number for a typical enterprise application, except when the switch serves as a tandem node in a network.

System calls

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

Total calls (T_{CALL}) = 0.5 × T_{CCS} × 100 ÷ WAHT

where weighted average holding time (WAHT) is given by:

WAHT = (R_I × AHT_{SS}) + (R_T × AHT_{TT}) + (I × AHT_{TS}) + (O × AHT_{ST})

and where AHT is the average holding time of a call in seconds. The subscript indicates where the call initiated from and terminates on, with S = set and T = trunk. For example, AHT_{ST} denotes that the call initiated from a set and terminates on a trunk.

Traffic equations and penetration factors

Total system calls comprise four different types of traffic:

1. Intraoffice calls (CSS) (set-to-set) (Intraoffice calls (CSS))

2. Tandem calls (CTT) (trunk-to-trunk) (Tandem calls (CTT))

3. Originating/outgoing calls (CST) (set-to-trunk) (Originating/outgoing calls (CST))

4. Terminating/incoming calls (CTS) (trunk-to-set) (Terminating/incoming calls (CTS))

1. Intraoffice calls (C_{SS})

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

    This parcel can be further broken down to three types:

    a. Intraoffice IP to IP calls (C_{2IP})
       = C_{SS} × P2 (require no DSP, no VT)

       pf1 = CSS × P2 × T_{CALL} = R_I × P2
       pf1 is the penetration factor for the intraoffice IP to IP calls

    b. Intraoffice IP to TDM set calls (C_{1IP})
       = C_{SS} × 2 × P × (1 − P) (require DSP)

       pf2 = C_{SS} × 2 × P × (1 − P) × T_{CALL} = 2 × R_I × P × (1 − P)
       pf2 is the penetration factor for the intraoffice IP to TDM set calls

    c. Intraoffice TDM set to TDM set calls (C_{NoIP})
       = C_{SS} × (1 − P)2 (require no DSP, no VT)

       pf3 = C_{SS} × (1 − P)2 ÷ T_{CALL} = R_I × (1 − P)2
       pf3 is the penetration factor for the intraoffice TDM to TDM calls

2. Tandem calls (C_{TT})

    = Total calls × Tandem ratio = T_{CALL} × R_T

The tandem calls can be further broken down into:

a. Tandem VT to TDM trunk calls (C_{T1VT})
   $= 2 \times$ Tandem VT calls $\times (1 - V)$
   $= 2 \times C_{TT} \times V \times (1 - V)$ (require DSP and VT)

   pf4 $= 2 \times C_{TT} \times V \times (1 - V) \div T_{CALL} = 2 \times R_T \times V \times (1 - V)$

b. Tandem TDM trunk to TDM trunk calls (C_{T2NoVT})
   $= C_{TT} \times (1 - V)^2$ (require no DSP, no VT)

   pf5 $= C_{TT} \times (1 - V)^2 \div T_{CALL} = R_T \times (1 - V)^2$

c. Tandem VT (H.323) to VT (SIP) calls (C_{T2HS})
   $= C_{TT} \times V^2 \times v_H \times v_S \times 2 \times 2$ (require VT)

   pf6 $= 4 \times C_{TT} \times V^2 \times v_H \times v_S \div T_{CALL} = 4 \times R_T \times V^2 \times v_H \times v_S$

   where v_H is the fraction of H.323 trunks to total VTs, and v_S is the fraction of SIP trunks to total VTs.

   > *Note:* If there is only one type of VT (either v_H or $v_S = 0$), the connection is handled at the Network Routing Service and no calls are offered to the Call Server. In this case, pf6 = 0.

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

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

   Originating/outgoing calls can be further broken down into:

   a. IP to VT calls (C_{STIV})
      $= C_{ST} \times$ (fraction of IP calls) $\times V$
      $= C_{ST} \times P \times V$ (require VT)

      pf7 $= C_{ST} \times P \times V \times T_{CALL} = O \times P \times V$

   b. IP to TDM trunk calls (C_{STID})
      $= C_{ST} \times$ (IP calls) $\times (1 - V)$
      $= C_{ST} \times P \times (1 - V)$ (require DSP)

      pf8 $= C_{ST} \times P \times (1 - V) \div T_{CALL} = O \times P \times (1 - V)$

   c. TDM set to VT calls (C_{STDV})
      $= C_{ST} \times (1 -$ fraction of IP calls$) \times V$
      $= C_{ST} \times (1 - P) \times V$ (require DSP, VT)

      pf9 $= C_{ST} \times (1 - P) \times V \div T_{CALL} = O \times (1 - P) \times V$

   d. TDM set to TDM trunk calls (C_{STDD})
      $= C_{ST} \times (1 -$ fraction of IP calls$) \times (1 - V)$
      $= C_{ST} \times (1 - P) \times (1 - V)$ (require no DSP, no VT)

      pf10 $= C_{ST} \times (1 - P) \times (1 - V) \div T_{CALL} = O \times (1 - P) \times (1 - V)$

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

= Total calls × Incoming ratio = T_{CALL} × I

Terminating/incoming calls can be further broken down into:

a. VT to TDM set calls (C_{TSVD})
   = C_{TS} × V × (1 − fraction of IP calls)
   = C_{TS} × V × (1 − P) (require DSP, VT)

   pf11 = C_{TS} × V × (1 − P) ÷ T_{CALL} = I × V × (1 − P)

b. VT to IP set calls (C_{TSVI})
   = C_{TS} × V × (fraction of IP calls)
   = C_{TS} × V × P (require VT)

   pf12 = C_{TS} × V × P ÷ T_{CALL} = I × V × P

c. TDM trunk to IP set calls (C_{TSDI})
   = C_{TS} × (1 − V) × (fraction of IP calls)
   = C_{TS} × (1 − V) × P (require DSP)

   pf13 = C_{TS} × (1 − V) × P ÷ T_{CALL} = I × (1 − V) × P

d. TDM trunk to TDM set calls (C_{TSDD})
   = C_{TS} × (1 − V) × (1 − fraction of IP calls)
   = C_{TS} × (1 − V) × (1 − P) (require no DSP, no VT)

   pf14 = C_{TS} × (1 − V) × (1 − P) ÷ T_{CALL} = I × (1 − V) × (1 − P)

Resource use equations

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

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

  C_{IP} = C_{2IP} + $C1_{IP}$ + C_{STIV} + C_{STID} + C_{TSVI} + C_{TSDI}

- Calls that require DSP resources:

  CDSP = C1IP + CT1VT + CSTID + CSTDV + CTSVD + CTSDI

- Calls that require Virtual Trunk resources:

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

  — Calls that require H.323 Virtual Trunks:

    HC_{VT} = C_{VT} × v_H

  — Calls that require SIP Virtual Trunks

    SC_{VT} = C_{VT} × v_S

Real-time calculations

This section describes the following real-time calculations:

- "Call Server utilization" (page 215)

- "Application and feature EBCs" (page 215)

- "Call Server real time" (page 216)

- "CPU real-time conversion for upgrades" (page 216)

The real time required to process a basic 2500-type set to 2500-type set call is an Equivalent Basic Call (EBC), the unit used to measure other, more complicated feature calls. Every feature call can be converted to EBCs by using its real-time factor (RTF).

RTF = (Real time of a feature call in ms ÷ Real time of a basic call) - 1

There are a total of 14 major combinations of set and trunk types of calls in the system. The real-time factor of each call type is denoted by f_i (i = 1 to 14). In addition, there are standard real-time factors for applications and features. Table 65 "Real-time factors" (page 213) provides the real-time factors.

Table 65
Real-time factors

Type of call	Real-time factor
Intraoffice calls:	
IP set to IP set (f_1)	0.86
IP set to TDM set (f_2)	1.70
TDM set to TDM set (f_3)	0.03
Tandem calls:	
Virtual Trunk to TDM trunk (f_4)	1.76
TDM trunk to TDM trunk (f_5)	1.76
H.323 Virtual Trunk to SIP Virtual Trunk (f_6)	1.73
Originating/outgoing calls:	
IP set to Virtual Trunk (f_7)	2.08
IP set to TDM trunk (f_8)	1.82
TDM set to Virtual Trunk (f_9)	1.71
TDM set to TDM trunk (f_{10})	0.69
Terminating/incoming calls:	
Virtual Trunk to TDM set (f_{11})	1.17
Virtual Trunk to IP set (f_{12})	1.39
TDM trunk to IP set (f_{13})	1.02
TDM trunk to TDM set (f_{14})	0.33
Application/feature calls:	
ACD agent without Symposium (f_{ACD})	0.13

Type of call	Real-time factor
Symposium (f_{SYM})	5.70
CallPilot (f_{CP})	1.70
Nortel Integrated Conference Bridge (f_{MICB})	1.59
Nortel Integrated Recorded Announcer (f_{MIRAN})	0.63
Nortel Integrated Call Assistant (f_{MICA})	0.57
Nortel Hospitality Integrated Voice Service (f_{MIVS})	0.57
Nortel Integrated Call Director (f_{MIPCD})	0.63
BRI ports (f_{BRI})	0.12
MDECT set (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
Microsoft Office factor (f_{MO})	2.33
Error term – minor feature overhead (f_{OVRH})	0.25

The real-time factor adjusts for the fact that a feature call generally requires more real time to process than a basic call. The impact on the system is a function of the frequency 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. Refer to for the equations to calculate penetration factors for the 14 major call types.

The real-time factors and penetration factors are used to generate the real-time multiplier (RTM), which in turn is used to calculate the overall system EBC.

The real-time multiplier is given by:

$$\text{RTM} = 1 + \text{Error\_term} + \sum_{i} (\text{Real-time factor fi} \times \text{Penetration factor pfi})$$

The Error_term accounts for features such as call transfer, three-way conference, call-forward-no-answer, and others that are difficult to single out to calculate real-time impact. The Error_term is usually assigned the value 0.25.

Call Server utilization

System real-time EBC = (Total system calls × Real-time multiplier) +
Application and feature EBCs

= (T_{CALL} × RTM) + Application and feature EBCs

Application and feature EBCs

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 66
Application and feature EBCs

Type	Calculation
ACD	ACD agents without Symposium + ACD agents with Symposium where ACD agents without Symposium = (1 − % Symposium) × f_{ACD} × (Number of IP ACD agents + number of TDM agents) × CCS per agent × 100 ÷ AHT_{AGT} and ACD agents with Symposium is user input. (If unknown, assume all ACD agent calls are with Symposium.)
Symposium	% Symposium × f_{SYM} × (Number of IP ACD agents + number of TDM agents) × CCS per agent × 100 ÷ AHT_{AGT}
CallPilot	(Number of Local CallPilot ports + number of Network CallPilot ports) × CCS × 100 ÷ AHT_{CP} × f_{CP}
Internal CDR	C_{SS} × f_{ICDR}
Incoming CDR	C_{TS} × f_{CCDR}
Outgoing CDR	C_{ST} × f_{OCDR}
Tandem CDR	C_{TT} × f_{TCDR}
Integrated Conference Bridge	Number of Integrated Conference Bridge ports × CCS × 100 ÷ AHT_{MICB} × f_{MICB}
Integrated Recorded Announcer	Number of Integrated Recorded Announcer ports × CCS × 100 ÷ AHT_{MIRAN} × f_{MIRAN}
Integrated Call Director	Number of Integrated Call Director ports × CCS × 100 ÷ AHT_{MIPCD} × f_{MIPCD}
Integrated Call Announcer	Number of Integrated Call Announcer ports × CCS × 100 ÷ AHT_{MICA} × f_{MICA}
Hospitality Integrated Voice Services	Number of Hospitality Integrated Voice Services ports × CCS × 100 ÷ AHT_{MIVS} × f_{MIVS}

Type	Calculation
BRI	# BRI ports $\times$ CCS $\times$ 100 $\div$ AHT_{BRI} $\times$ f_{BRI}
MDECT	L_{DECT} $\times$ 100 $\div$ WAHT $\times$ f_{DECT}
CPND	$(C_{1IP} + C_{NoIP} + C_{TSVD} + C_{TSDD})$ $\times$ f_{CPND}
Converged Desktop (CD)	$(C_{SS} \times 0.1 + C_{TT} + C_{ST} + C_{TS})$ $\times$ r_{DTP} $\times$ f_{DTP}
SIP CTI/TR87 (MO)	$(C_{SS} \times 0.1 + C_{TT} + C_{ST} + C_{TS})$ $\times$ r_{MO} $\times$ f_{MO}

Call Server real time

Compare the system EBC with the Meridian 1/CS 1000M CPU rated capacity to determine the processor utilization.

CPU utilization = System real-time EBC $\div$ Rated capacity of processor ($\times$ 100 to get a percentage)

The rated capacity of the SSC in the Small System is 35 000 EBC.

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 $\times$ [1 + (SWRC $\div$ 100)] $\times$ CPTU

where

CRTU = CPU loading from the existing switch converted to an equivalent load on the new processor, in percent.
RTU = Current CPU usage, in percent (from the TFS004 report of the existing switch).
SWRC = Software release degradation factor, in percent. Since every new release is enhanced with new features and capabilities, the processing power of the existing CPU is degraded to some extent (typically 10-20%) by the newer release.

CPTU = Capacity ratio of the existing CPU to the new CPU. The ratio is always less than 1 (unless the same CPU is used, in which case it is equal to 1).

If CRTU > CPTU, 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 67 "Software release degradation factors (SWRC)" (page 217) lists the software release degradation factors for supported software upgrades.

Table 67
Software release degradation factors (SWRC)

	Degradation factor (%)	
From	**To Succession 3.0 Software**	**To CS 1000 Release 4.5**
Rls 18	151	188
Rls 19	139	174
Rls 20B	96	125
Rls 21B	78	104
Rls 22	53	75
Rls 23	42	62
Rls 23C	38	58
Rls 24B	12	28
Rls 25B	3	18
SR2	1	16
SR3	–	14
SR4	–	9

Table 68 "Ratio of existing processor capacity to SSC capacity (CPTU)" (page 217) gives capacity ratio values for supported processor upgrades.

Table 68
Ratio of existing processor capacity to SSC capacity (CPTU)

From CPU type	**EBC ratio**
11	0.08
11E	0.10
11C	1.00
21E	0.27

Example   To convert the loading of an Option 11C using CPU 21E and running Rls 22 to upgrade to an SSC card and CS 1000 Release 4.5 software, where the TFS004 reading is 80%:

SWRC = 61 and CPTU = 0.27
CRTU = 80 × [1 + (61 × 100)] × 0.27 = 34.8%

The new system should reserve 34.8% to handle existing traffic. The remaining 65.2% capacity can be engineered for future growth.

Note: Nortel recommends always keeping some spare capacity as a safety margin.

DSP/Media Card calculations

DSP resources are provided by Media Cards. The total DSP/Media Card requirement is the sum of DSP requirements for various functions, which are calculated separately.

- "DSP ports for Conference" (page 219)
- "DSP ports for general traffic" (page 220)
- "DSP ports for major applications" (page 221)
- "Special ACD treatment for non-blocking access to DSP ports" (page 221)
- "Total DSP requirements" (page 223)
  - "General configuration (ACD agent sets less than 15 percent of total sets)" (page 223)
  - "Call center application (ACD agent sets greater than 15 percent of total sets)" (page 223)

For reasons explained in the ""System capacities" (page 153)" chapter (see "Traffic capacity engineering algorithms" (page 181)), 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 69 "Erlang B and Poisson values, in 32-port increments" (page 219) provides Erlang B and Poisson values for P.01 Grade-of-Service (GoS) in 32-port increments. The DSP resource required to handle the offered traffic is the number of ports corresponding to the

first Erlang B CCS capacity greater than the calculated traffic value. The Poisson values are used to calculate Virtual Trunk requirements (see "Virtual Trunk calculations" (page 223)).

Table 69
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 69 "Erlang B and Poisson values, in 32-port increments" (page 219)) × (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 Conference

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

Applications are another source of demand for the conference feature. Conference usage for Integrated Conference Bridge is treated separately, as part of the calculations for application ports. For other applications, the default is two conference loops, with a total of 60 channels, per network group. If a particular application requires a different number of conference ports, use the specific number.

The equation to calculate the number of DSP ports the system requires for Conference is:

Equation 1
Number of DSP ports for Conference = (Total number of sets) × P × r_{Con} × 0.4

where r_{Con} is the ratio of conference loops to traffic loops. The default value of r_{Con} is 0.07 because, for each network group, there are assumed to be 2 conference loops and 28 traffic loops ($r_{Con} = 2 \div 28 = 0.07$). 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 sets 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).

DSP ports for general traffic

There are three steps to calculate the number of DSP ports required for general traffic:

Step	Action
1	Calculate the number of calls that require DSP resources.
	DSP calls (C_{DSP}) = Intraoffice IP-TDM set calls (C_{1IP}) + Tandem VT-TDM trunk calls (C_{T1VT}) + IP-TDM trunk calls (C_{STID}) + TDM set-VT calls (C_{STDV}) VT-TDM set calls (C_{TSVD}) + TDM-IP set calls (C_{TSDI})
	= $C_{1IP} + C_{T1VT} + C_{STID} + C_{STDV} + C_{TSVD} + C_{TSDI}$
	For sites where the proportion of ACD agent sets is less than 15% of the total sets in the system, C_{DSP} includes all general traffic seeking DSP service.
	Sites where the proportion of ACD agent sets exceeds 15% of the total sets in the system are considered to be call centers. For call centers, CDSP is a reduced total that excludes ACD CCS. See "Special ACD treatment for non-blocking access to DSP ports" (page 221) and Note 3 on 3.
2	Convert DSP calls to CCS.
	DSP CCS = $C_{DSP} \times$ WAHT $\div$ 100
3	Using the Erlang B table for P.01 GoS (see Table 69 "Erlang B and Poisson values, in 32-port increments" (page 219)), find the corresponding number of DSP ports required.

—End—

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).

Table 70 "DSP port requirements for applications" (page 221) provides the equations to calculate the number of DSP ports required for each application.

Table 70
DSP port requirements for applications

Application or port type	Calculation
Integrated Recorded Announcer	Number of Integrated Recorded Announcer ports × P
Integrated Conference Bridge	Number of Integrated Conference Bridge ports × P
Integrated Call Director	Number of Integrated Call Director ports × P
Integrated Call Assistant	Number of Integrated Call Assistant ports × P
Hospitality Integrated Voice Service	Number of Hospitality Integrated Voice Service ports × P
BRI	Number of SILC ports × P = Number of BRI users × 2 × P
CallPilot ports	(Number of local CallPilot ports × P) + (Number of network CallPilot ports × P) (see Note)
Agent Greeting ports	Number of Agent Greeting ports × P
Note: 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 non-blocking access to DSP ports

The following section applies to call centers, which are defined as sites where the number of ACD agent sets exceeds 15% of the total sets 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.

Therefore, use the fixed rule of one DSP port for each ACD agent set requiring a DSP resource, in order to provide non-blocking access between an ACD agent set and a DSP. ACD agent sets require DSP resources only when calls are coming from TDM trunks to IP agent sets or from Virtual Trunks to TDM agent sets.

In general, Media Cards are system resources that are available to all traffic sources, including ACD agent sets and regular phones. Zoning control is the only way to provide non-blocking access to DSP ports for ACD agent sets 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 sets, 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 sets. The steps are as follows:

Step	Action
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 sets) × 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 64 "Major parameters for VoIP resource calculations" (page 206) for the equations to calculate the ratios. See also Note 3.)
3	Use the reduced system CCS and new ratios to calculate calls requiring DSP and Virtual Trunk resources. (See "Traffic equations and penetration factors" (page 210) 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 69 "Erlang B and Poisson values, in 32-port increments" (page 219)), 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 sets. A DSP port is needed only when calls are coming from TDM trunks (ratio $1 - V$) to IP agent sets or from Virtual Trunks (ratio V) to TDM agent sets.

Equation 4

Number of DSP ports = (Number of IP ACD agent sets) $\times$ ($1 - V$) + (Number of TDM ACD agent sets) $\times$ V

—End—

Total DSP requirements
General configuration (ACD agent sets less than 15 percent of total sets)
Total number of DSP ports = Equation 1 + Equation 2 + Equation 3

Call center application (ACD agent sets greater than 15 percent of total sets)
Total number of DSP ports = Equation 1 + Equation 2a + Equation 3 + Equation 4

Virtual Trunk calculations

For reasons explained in the ""System capacities" (page 153)" chapter (see "Traffic capacity engineering algorithms" (page 181)), the Poisson model is used to calculate trunk requirements.

Table 69 "Erlang B and Poisson values, in 32-port increments" (page 219) 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) $\times$ (CCS from Table 69 "Erlang B and Poisson values, in 32-port increments" (page 219)) $\times$ (Number of access ports for table CCS)

Perform the following steps to calculate the number of access ports required:

Step	Action
1	Estimate the Virtual Trunk requirement by adding together all the calls that require the service of access ports.
	Virtual Trunk calls (C_{VT}) = Tandem VT-TDM trunk calls (C_{T1VT}) + IP-VT calls (C_{STIV}) + TDM set-VT calls (C_{STDV}) + VT-TDM set calls (C_{TSVD}) + VT-IP set calls (C_{TSVI}) + H.323-SIP VT calls (C_{T2HS})
	= $C_{T1VT} + C_{STIV} + C_{STDV} + C_{TSVD} + C_{TSVI} + C_{T2HS}$
	For sites where the proportion of ACD agent sets is less than 15% of the total sets in the system, C_{VT} includes all general traffic seeking an access port.
	Sites where the proportion of ACD agent sets exceeds 15% of the total sets in the system are considered to be call centers. For call centers, CVT is a reduced total that excludes ACD CCS. See "Special ACD treatment for non-blocking access to DSP ports" (page 221) and Note 3.
2	Convert Virtual Trunk calls to CCS.
	Virtual Trunk CCS (VT_{CCS}) = C_{VT} × WAHT ÷ 100
3	For call centers, since the calculated Virtual Trunk calls exclude ACD traffic, restore ACD traffic so that the final number of Virtual Trunks will be sufficient to handle both general and ACD traffic.
	Final Virtual Trunk CCS = (Calculated VT_{CCS} without ACD) + [(Number of IP ACD agent sets) + (Number of TDM ACD agent sets)] × V × (CCS per ACD agent) ÷ (CCS per trunk)
	where:
	default CCS per ACD agent = 33 CCS default CCS per trunk = 28 CCS
	The expanded Virtual Trunk CCS is inflated by the ratio of 33/28 to reflect the fact that more Virtual Trunks are needed to carry each agent CCS. This is because the traffic levels engineered for ACD agents and Virtual Trunks are different.
4	Use the SIP and H.323 ratios to determine how the Virtual Trunk access ports will be allocated to the two groups.
	SIP Virtual Trunk CCS (SVT_{CCS}) = VT_{CCS} × v_S H.323 Virtual Trunk CCS (HVT_{CCS}) = VT_{CCS} × v_H
5	Using the Poisson table for P.01 GoS (see Table 69 "Erlang B and Poisson values, in 32-port increments" (page 219) or "Trunk traffic Erlang B with P.01 Grade-of-Service" (page 340)), find the corresponding number of SIP and H.323 access ports required.

Note: 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.

—End—

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 greater, 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)" (page 236).

A discrepancy between calculated and input Virtual Trunks is significant because system resources such as DSP ports and Virtual Trunk licenses depend on the accuracy of the traffic split. Imbalanced Virtual Trunk traffic renders the resulting equipment recommendation unreliable.

For example, if the calculated number of Virtual Trunks is 80 but the original input value was 60, and the user decides to use the original input value of 60 to calculate bandwidth and Signaling Server requirements, the resulting system will likely provide service inferior to the normal expected P.01 GoS. On the other hand, if the user input was 80 and the calculated result is 60, the user must 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:

$$VT \text{ erlangs} = VT_{CCS} \div 36$$

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. Refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)* for the bandwidth table and for more information about calculating LAN/WAN bandwidth requirements.

Signaling Server algorithm

The Signaling Server algorithm in the NNEC 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 added to determine the total Signaling Server requirement.

In most cases, the individual calculations divide the configuration's requirement for an applicable parameter (endpoint, call, set, trunk) into the system limit for that parameter. The particular application's Signaling Server requirement is determined by the parameter with the largest proportional resource requirement, adjusted for redundancy.

Table 71 "Signaling Server algorithm constant and variable definitions"
(page 227) defines the variables used in the calculations.

Table 71
Signaling Server algorithm constant and variable definitions

Algorithm term	Description	Value	Notes
NRA	Network Routing Service (NRS) Alternate required	enter (see Note 2)	Yes or No.
NRC	NRS calls per hour	enter (see Note 2)	Two components (one local, one network): $NRC = NRC_0 + NRC_{NET}$
NRC_{HL}	NRS calls per hour	100 000 (see Note 1)	Hardware limit for Signaling Server.
NRD	NRS CDP + UDP entries	enter (see Note 2)	
NRD_1	NRS CDP + UDP entries limit	20 000 (see Note 4)	Software limit.
NRE	NRS endpoints	enter (see Note 2)	(= 0 if NRS, which is a network-wide resource, is not provisioned in this node)
NRE_1	NRS endpoints limit	5000 (see Note 4)	Software limit.
NRP	NRS product of endpoint and CDP/UDP entries	- (see Note 3)	Interim calculation.
NRP_{SL}	NRS product of endpoint and CDP/UDP entries	20 000 (see Note 4)	Software limit.
GSA	SIP Gateway Alternate required	enter (see Note 2)	Yes or No.
GWA	H.323 Gateway Alternate required	enter (see Note 2)	Yes or No.
C_{IP}	IP Phones calls per hour	enter/ derived (see Note 2)	Busy Hour calls from all IP Phones.

Note 1: Constant in the formulas.

Note 2: Variable to be entered into the formula.

Note 3: Constant that will update with platform changes.

Note 4: Constant that will update with system software releases.

Note 5: Calculated result.

Algorithm term	Description	Value	Notes
IPC_{HL}	IP Phones calls per hour limit	15 000 (see Note 4)	Hardware limit.
IPL	IP Phones	enter (see Note 2)	
IPL_{DB}	IP set limit with Personal Directory, Callers List, and Redial List database	1000 (see Note 1)	IP Phone limit per Signaling Server reduced due to Personal Directory, Callers List, and Redial List database.
IPL_{SL}	IP Phones limit	5000 (see Note 4)	Software limit.
NRD_{HL}	NRS product of endpoint and CDP/UDP entries	20 000 (see Note 1)	Hardware limit.
SSNR	NRS Signaling Server calculation	calc (see Note 5)	Real number requirement (for example, 1.5) (= 0 if NRS is not provisioned in this node)
SSGW	NRS Signaling Server requirements	calc (see Note 5)	Whole number requirement including Alternate.
SSHR	H.323 Gateway Signaling Server calculation	calc (see Note 5)	Real number requirement (for example,1.5).
SSHW	H.323 Gateway Signaling Server requirements	calc (see Note 5)	Whole number requirement including Alternate.
SST	Total count of Signaling Servers required	calc (see Note 5)	
SSTR	TPS Signaling Server calculation	calc (see Note 5)	Real number requirement (for example,1.5).
SSTW	TPS Signaling Server requirements	calc (see Note 5)	Whole number requirement including Alternate.
TPSA	TPS N+1 redundancy required	enter (see Note 2)	Yes or No.

Note 1: Constant in the formulas.

Note 2: Variable to be entered into the formula.

Note 3: Constant that will update with platform changes.

Note 4: Constant that will update with system software releases.

Note 5: Calculated result.

Algorithm term	Description	Value	Notes
HVTC$_{HL}$	H.323 Gateway calls per hour limit	18 000 (see Note 4)	Hardware limit. HVTC$_{HL}$ = v_H × C$_{VT}$
SVTC$_{HL}$	SIP Gateway calls per hour limit	27 000 (see Note 4)	Hardware limit. SVTC$_{HL}$ = v_S × C$_{VT}$
VT$_{SIP}$	SIP Gateway access ports per Signaling Server	1800 (see Note 4)	CPU limit.
VT$_{323}$	H.323 Gateway access ports per Signaling Server	1200 (see Note 4)	CPU limit.
TR87			
TR87CL			
TR87A			

Note 1: Constant in the formulas.

Note 2: Variable to be entered into the formula.

Note 3: Constant that will update with platform changes.

Note 4: Constant that will update with system software releases.

Note 5: Calculated result.

Signaling Server calculations

All the Signaling Server software components can function either on shared or on stand-alone Signaling Servers. System traffic and user requirements (for alternate, redundant, or dedicated Signaling Servers) determine how many Signaling Servers will be required. The Signaling Server algorithm takes account of all these requirements by performing the following calculations in sequence:

1. Signaling Server for Personal Directory, Callers List, and Redial List database (SSDB)

2. Network Routing Service calculation (SSNR)

3. Terminal Proxy Server calculation (SSTR)

4. H.323 Gateway calculation (SSHR)

5. SIP Gateway calculation (SSSR)

6. Total Signaling Server requirements (SST)

1. **Signaling Server for Personal Directory, Callers List, and Redial List database (SSDB)**

   Personal Directory, Callers List, and Redial List (PD/CL/RL) calculations assume that the database resides either on a stand-alone Signaling Server or on a Signaling Server shared with all the other applications. This assumption simplifies the engineering and provisioning rules.

   $SSDB$ = a if no PD/CL/RL feature

   = b if yes on feature, and sharing functions on Signaling Server

   = c if yes on feature, and a stand-alone database Signaling Server

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

   SSNR = larger of:

   {

   a $NRE \div NRE_I$ endpoints (software limit)

   b $NRD \div NRD_I$ dial plan entries (software limit)

   c $NRC \div NRC_{HL}$ calls per hour (hardware limit)

   }

   NRC can be a hardware, CPU, or memory limit. It includes local calls (NRC_0) and network Virtual Trunks (VT_{NET}) for this Network Routing Service.
   NRC_0 is obtained from the main switch calculation.
   $NRC_{NET} = VT_{NET} \times (CCS \text{ per } VT) \times 100 \div WAHT \div 2$
   $NRC = NRC_0 + NRC_{NET}$

   The calculation for NRC_{NET} requires converting both VT_{323} and VT_{SIP} (from user input) to H.323 and SIP calls. The Signaling Server's capacity for handling SIP calls is different from its capacity for H.323 calls.

   Therefore, H.323 calls are converted to SIP calls before the load on the Signaling Server is calculated.

   Convert H.323 calls to SIP calls by using the ratio of the real-time factors for calls from IP sets to SIP and H.323 Virtual Trunks:
   $f_{H/S}$ = (H.323 call real time) $\div$ (SIP call real time)

   Equation (c) in the SSNR calculation becomes:

   = $[NRC_S + (f_{H/S} \times NRC_H)] \div NRC_{HLS}$
   where:
   NRC_S = the sum of local and network SIP calls the NRS is handling
   NRC_H = the sum of local and network H.323 calls the NRS is handling

NRC_{HLS} = the Signaling Server's capacity for handling SIP calls

Equation (c) represents the loading of the Signaling Server for handling NRS calls. It is compared with equations (a) and (b) in order to determine the highest of all potential usages.

If the user wants the Network Routing Service in a dedicated Signaling Server, round up SSNR before proceeding with further calculations:

SSNW = ROUNDUP(SSNR) × NRA (= 2 if true; else = 1)
where NRA = if Alternate NRS is needed.

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

   SSTR = larger of:

   {

a	If SSDB = a or c, $IPL \div IPL_{SL}$	no PD/CL/RL or sharing IP Phones limit
b	If SSDB = b,	with PD/CL/RL and sharing
	If IPL <= IPL_{DB}, $IPL \div IPL_{DB}$	
	If IPL > IPLDB, $1 + [(IPL - IPL_{DB}) \div IPL_{SL}]$	database platform limit (1000 for the first Signaling Server)
c	$IPC \div IPC_{HL}$	calls per hour limit

   }

   If the user wants Terminal Proxy Server(s) in a dedicated Signaling Server, round up SSTR before proceeding with further calculations:

   SSTW = ROUNDUP(SSTR) + TPSA (= 1 if true; else = 0)
   where TPSA = if N+1 redundant TPS is needed.

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

   SSHR = larger of:

   {

a	$HVT \div HVT_{SL}$	number of trunks (software limit)
b	$C_{VT} \div HVTC_{HL}$	calls per hour (hardware limit)

   }

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

   SSHW = ROUNDUP(SSHR) × GWA (= 2 if true; else = 1)
   where GWA = if Alternate H.323 Gateway is needed.

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

SSSR = larger of:

{

a $SVT \div SVT_{SL}$ number of trunks (software limit)

b $C_{VT} \div SVTC_{HL}$ calls per hour (hardware limit)

}

If the user wants SIP Gateway(s) in a dedicated Signaling Server, round up SSSR before proceeding with further calculations:

SSSW = ROUNDUP(SSSR) × GSA (= 2 if true; else = 1)
where GSA = if Alternate SIP Gateway is needed.

6. **Total Signaling Server requirements (SST)**

The final calculation of SST requires picking the formula that suits the particular configuration and user input.

SST = evaluate in order,

a   If (SSNR + SSTR + SSHR + SSSR) < 1

SST = ROUNDUP(SSNR + SSTR + SSHR + SSSR) + (1 if NRA, GWA, GSA, or TPSA true; else 0) + (1 if SSDB = c; else 0)

OR

b   If (SSTR + SSHR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1

SST = SSNW + [ROUNDUP(SSTR + SSHR + SSSR) × (2 if GWA, GSA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)

OR

c   If (SSNR + SSHR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1

SST = SSTW + [ROUNDUP(SSNR + SSHR + SSSR) × (2 if NRA, GWA, or GSA true; else 1)] + (1 if SSDB = c; else 0)

Note: When the database server is not stand-alone, it is shared with the TPS server. The TPSA calculation dictates whether an additional Signaling Server is needed.

OR

d   If (SSTR + SSNR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1

SST = SSHW + [ROUNDUP(SSTR + SSNR + SSSR) × (2 if NRA, GSA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)

OR

e   If (SSTR + SSNR + SSHR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1

SST = SSSW + [ROUNDUP(SSTR + SSNR + SSHR) × (2 if NRA, GWA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)

OR

Note: If the process reaches this step, it means that (SSGR + SSTR + SSHR + SSSR) > 1 and also that there is no combination of three functions that can share one Signaling Server. The algorithm continues by evaluating whether there are combinations of two functions that can share a Signaling Server.

f   If (SSNR + SSTR) < 1 and (SSHR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR + SSTR) × (2 if NRA or TPSA true; else 1)] + [ROUNDUP(SSHR + SSSR) × (2 if GWA or GSA true; else 1)] + (1 if SSDB = c; else 0)

OR

g   If (SSNR + SSHR) < 1 and (SSTR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR + SSHR) × (2 if NRA or GWA true; else 1)] + [ROUNDUP(SSTR + SSSR) × (2 if TPSA or GSA true; else 1)] + (1 if SSDB = c; else 0)

OR

h   If (SSNR + SSSR) < 1 and (SSTR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR + SSSR) × (2 if NRA or GSA true; else 1)] + [ROUNDUP(SSTR + SSHR) × (2 if TPSA or GWA true; else 1)] + (1 if SSDB = c; else 0)

OR

i   If (SSTR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR) × (2 if NRA true; else 1)] + [ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSTR + SSHR) × (2 if TPSA or GWA true; else 1)] + (1 if SSDB = c; else 0)

OR

j   If (SSNR + SSTR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSHR) × (2 if GWA true; else 1)] +
[ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSNR +
SSTR) × (2 if NRA or TPSA true; else 1)] + (1 if SSDB = c; else 0)

OR

k   If (SSNR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSTR) + (1 if TPSA true; else 0)] +
[ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSNR +
SSHR) × (2 if NRA or GWA true; else 1)] + (1 if SSDB = c; else 0)

OR

l   If (SSNR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSTR) + (1 if TPSA true; else 0)] +
[ROUNDUP(SSHR) × (2 if GWA true; else 1)] + [ROUNDUP(SSNR +
SSSR) × (2 if NRA or GSA true; else 1)] + (1 if SSDB = c; else 0)

OR

m   If (SSTR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR) × (2 if NRA true; else 1)] +
[ROUNDUP(SSHR) × (2 if GWA true; else 1)] + [ROUNDUP(SSTR +
SSSR) × (2 if TPSA or GSA true; else 1)] + (1 if SSDB = c; else 0)

OR

n   If (SSHR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1

SST = [ROUNDUP(SSNR) × (2 if NRA true; else 1] + ROUNDUP(SSTR)
+ (1 if TPSA true; else 0) + [ROUNDUP(SSHR + SSSR) × (2 if GWA
or GSA true; else 1)] + (1 if SSDB = c; else 0)

If the scenario has not fallen into any of the above 14 cases, it is not possible
to share Signaling Server functions. Use the following equation to calculate
the total Signaling Server requirement:

SST = ROUNDUP(SSNW + SSTW + SSHW + SSSW) + (1 if SSDB
= c; else 0)

where

SSNW = SSNR(ROUNDUP if dedicated) + [ROUNDUP(SSNR) × (1
if NRA true; else 0)]
SSTW = SSTR(ROUNDUP if dedicated) + (1 if TPSA true; else 0)

`SSHW` = SSHR(ROUNDUP if dedicated) + [ROUNDUP(SSHR) × (1 if GWA true; else 0)]
`SSSW` = SSSR(ROUNDUP if dedicated) + [ROUNDUP(SSSR) × (1 if GSA true; else 0)]

> *Note:* It is possible that the sum of two or three functions will be greater than 1, but their fractional parts may still be able to share a Signaling Server with the last function's fraction. In order to avoid overstating an individual function's needs and over-provisioning the total requirement, round off the Signaling Server requirement to a higher integer only after the fraction portion of all functions has been summed.

Refer to for a numerical example illustrating the algorithm.

Maximum number of Failsafe Network Routing Services

This algorithm defines the maximum number of Failsafe Network Routing Services (RSF) that can be configured. The maximum RSF is limited by the Primary Network Routing Service (RSP) configuration.

RSF is less than or equal to RSPE
$RSF = (RDE_L \div RSPE) \times [FR - (RFR_S \text{ or } RFR_C)] \times (DDR \div 24) \times (RSP_C)$

Simplified formulas:

$RSF = (16\ 000 \div RSPE)$ for stand-alone Network Routing Service
$RSF = (10\ 000 \div RSPE)$ for collocated Network Routing Service

Table 72 "RSF algorithm constant and variable definitions" (page 235) 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 (see Note 3)	In one day, the minimum number of synchronizations of dynamic data from Active RD to a RSF.

Note 1: Constant in the formulas.

Note 2: Variable to be entered into the formula.

Note 3: Constant that will update with platform changes.

Note 4: Constant that will update with system software releases.

Note 5: Calculated result.

Algorithm term	Description	Value	Notes
FR	FTP Resource	10 (see Note 4)	Software limit.
RDE_L	NRS endpoints limit	5000 (see Note 4)	Software limit.
RSF	Maximum Failsafe NRS allowed	calc (see Note 5)	
RSP_C	RSP CPU performance	1.0 (see Note 3)	PIII 700 MHz; 512 MByte; 20 GByte
RSPE	RSP endpoints	enter (see Note 2)	
RFR_C	Reserved FTP Resource Collocated	5 (see Note 4)	Software limit. RSP shares Signaling Server with other applications, such as TPS. Reserve 3 for other applications.
RFR_S	Reserved FTP Resource Standalone	2 (see Note 4)	Software limit. RSP is only application on Signaling Server. Reserve 1 for Static updates and 1 spare.

Note 1: Constant in the formulas.

Note 2: Variable to be entered into the formula.

Note 3: Constant that will update with platform changes.

Note 4: Constant that will update with system software releases.

Note 5: Calculated result.

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 NNEC tool will generate a warning message indicating traffic imbalance. The user may want to change the input data and rerun the calculation.

Two types of imbalances can occur

- "Virtual Trunks" (page 236)
- "Line and trunk traffic" (page 237)

Virtual Trunks

When the VT number input by the user differs significantly from the calculated VT number (more than 20% difference), the NNEC tool uses the calculated number and reruns the algorithm to obtain a newer VT number. If

the resulting VT number is not stable (in other words, after each rerun, a new calculated VT number is obtained), the program repeats the calculation cycle up to six times. This recalculation looping is built into the NNEC 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 pre-engineering 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 NNEC 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). 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 NNEC 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.

$$\text{Calculated line CCS } (LC_{CCS}) = (C_{SS} + C_{ST} + C_{TS}) \times WAHT \div 100$$

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}) × WAHT ÷ 100

The difference between T_{TCCS} and TC_{CCS} is the imbalanced trunk CCS.

Because the calculated CCS factor in traffic ratios, line and trunk CCS can be significantly imbalanced if these ratios are inconsistent. For example, if the intraoffice, incoming, and outgoing ratios are based on contradictory assumptions, the calculated line CCS may be much higher than the number of trunks can absorb.

Table 73 "Tips to reduce traffic imbalances" (page 238) 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 set or number of sets. • 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 difficult 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.

Note: These parameter values are typical for systems in operation, but the values for the ratios are not the defaults.

- Intraoffice ratio (RI): 0.15

- Tandem ratio (RT): 0.02

- Incoming ratio (I): 0.60

- Outgoing ratio (O): 0.23

  In fraction of calls, the above ratios add up to 1.

- AHT_{SS} = 60 [average hold time (AHT) for set to set ($_{SS}$)]

- AHT_{TS} = 150 [AHT for trunk to set ($_{TS}$)]

- AHT_{ST} = 150 [AHT for set to trunk ($_{ST}$)]

- AHT_{TT} = 180 [AHT for trunk to trunk ($_{TT}$)]

Given configuration

A CS 1000M Small Systemwith the following configuration data:

- 100 digital and analog sets at 5 CCS/set

- 450 IP sets at 5 CCS/IP set

  — including 2 attendant consoles (equivalent to ACD agents) with digital sets at 33 CCS/agent

- 100 trunks

  — including 68 Virtual Trunks (20H.323 and 48SIP) at 26 CCS/trunk (The numbers for H.323 and SIP Virtual Trunks are input from user response to a GUI request in the NNEC.)

- Network Virtual Trunks served by this Gatekeeper: 400 (This is another input from the user interface.)

- 24 local CallPilot ports at 26 CCS/CP port

- 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.

  — Integrated Recorded Announcer ports: 12 (HT = 90)

  — Integrated Call Assistant ports: 8 (HT = 180)

  — Integrated Call Director ports: 6 (HT = 60)

- Features with processing overhead but no hardware ports:

  — CPND percentage: 20% of TDM set calls

  — Converged Desktop percentage: 10% 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: No

  — Outgoing CDR: Yes

  — Tandem CDR: No

  — Symposium-processed ACD calls: 0%

  — ACD calls without Symposium: 100%

Real-time factors are based on Table 65 "Real-time factors" (page 213).

Calculations

Note: The calculations in this example were performed by spreadsheet. Some rounding off may have occurred.

- The ACD agent to total set ratio = 2 ÷ (100 + 450) = 0.36% This ratio is less than the 15% threshold, so the site is not considered a call center. All ACD traffic is included in call distribution calculations. Refer to "DSP ports for general traffic" (page 220) for more information.

- TDM sets CCS = 100 × 5 = 500 CCS

- IP sets CCS = [(450 − 2) × 5] + (2 × 33) = 2306 CCS

- Fraction of IP calls (P) = 2306 ÷ (500 + 2306) = 0.82

- Weighted average holding time (WAHT)
  = (60 × 0.15) + (150 × 0.60) + (150 × 0.23) + (180 × 0.02) = 137 seconds

- Total line CCS (LCCS) = 500 + 2306 = 2806

- 100 trunks at 26 CCS per trunk:

  Fraction of Virtual Trunks (V) = 68 ÷ 100 = 0.68
  Virtual Trunk traffic (VTCCS) = 68 × 26 = 1768
  TDM trunk CCS (TTDM) = (100 − 68) × 26 = 832
  vH = 20 ÷ (20 + 48) = 0.29
  vS = 48 ÷ (20 + 48) = 0.71

  Total trunk CCS (TTCCS) = 1768 + 832 = 2600

- Total CCS (TCCS) = 2806 + 2600 = 5406

- Total calls (TCALL) = 0.5 × TCCS × 100 ÷ WAHT
  = 0.5 × 5406 × 100 ÷ 137 = 1973

- The system calls are comprised of four different types of traffic: Intraoffice calls (set-to-set) (CSS); Tandem calls (trunk-to-trunk) (CTT); Originating/Outgoing calls (set-to-trunk) (CST); Terminating/Incoming calls (trunk-to-set) (CTS).

  1. Intraoffice calls (CSS) = TCALL × RI
     = 1973 × 0.15 = 296

     a. Intraoffice IP to IP calls (C2IP) = CSS × P2
        = 296 × 0.82 × 0.82 = 200
        (require no DSP, no VT)
        pf1 = 200 ÷ 1973 = 0.10

     b. Intraoffice IP to TDM calls (C1IP) = CSS × 2 × P × (1 − P)
        = 296 × 2 × 0.82 × (1 − 0.82) = 87
        (require DSP)
        pf2 = 87 ÷ 1973 = 0.04

     c. Intraoffice TDM to TDM (CNoIP) = CSS × (1 − P)2
        = 296 × (1 − 0.82) × (1 − 0.82) = 9
        (require no DSP, no VT)
        pf3 = 9 ÷ 1973 = 0.005

  2. Tandem calls (CTT) = TCALL × RT
     = 1973 × 0.02 = 39 calls

     a. Tandem VT to TDM calls (CT1VT) = 2 × CTT × V × (1 − V) = 2 × 39 × 0.68 × (1 − 0.68) = 17
        (require DSP and VT)
        pf4 = 17 ÷ 1973 = 0.01

     b. Tandem TDM to TDM calls (CT2NoVT) = CTT × (1 − V) × (1 − V)
        = 39 × (1 − 0.68) × (1 − 0.68) = 4
        (require no DSP, no VT)
        pf5 = 4 ÷ 1973 = 0

     c. Tandem VT (H.323) to VT (SIP) calls (CT2HS)
        = CTT × V2 × vH × vS × 2 × 2
        = 39 × 0.68 × 0.68 × 0.29 × 0.71 × 4 = 15
        (require no DSP, VT)
        pf6 = 15 ÷ 1973 = 0.008

  3. Originating/outgoing calls (CST) = TCALL × O
     = 1973 × 0.23 = 454 calls

     a. IP to VT calls (CSTDI) = CST × P × V
        = 454 × 0.82 × 0.68 = 254
        (require VT)
        pf7 = 254 ÷ 1973 = 0.13

     b. IP to TDM calls (CSTID) = CST × P × (1 − V)

= 454 × 0.82 × (1 − 0.68) = 119
(require DSP)
pf8 = 119 ÷ 1973 = 0.06

c. TDM to VT calls (CSTDV) = CST × (1 − P) × (V)
= 454 × (1 − 0.82) × 0.68 = 55
(require DSP, VT)
pf9 = 55 ÷ 1973 = 0.03

d. TDM to TDM calls (CSTDD) = CST × (1 − P) × (1 − V)
= 454 × (1 − 0.82) × (1 − 0.68) = 26
(require no DSP, no VT)
pf10 = 26 ÷ 1973 = 0.01

4. Terminating/incoming calls (CTS) = TCALL × I = 1973 × 0.60 = 1184 calls

a. VT to TDM calls (CTSVD) = CTS × V × (1 − P)
= 1184 × 0.68 × (1 − 0.82) = 143
(require DSP, VT)
pf11 = 143 ÷ 1973 = 0.07

b. VT to IP calls (CTSVI) = CTS × V × P
= 1184 × 0.68 × 0.82 = 662
(require VT)
pf12 = 662 ÷ 1973 = 0.34

c. TDM to IP calls (CTSDI) = CTS × (1 − V) × P
= 1184 × (1 − 0.68) × 0.82 = 311
(require DSP)
Pf13 = 311 ÷ 1973 = 0.16

d. TDM to TDM calls (CTSDD) = CTS × (1 − V) × (1 − P)
= 1184 × (1 − 0.68) × (1 − 0.82) = 68
(require no DSP, no VT)
pf14 = 68 ÷ 1973 = 0.03

- From the above data, the real-time multiplier can be obtained:

  Real-time multiplier per call
  = 1 + (f1 × pf1) + (f2 × pf2) + (f3 × pf3) + ... + (f14 × pf14) + Error_term = 1 + (0.86 × 0.10) + (1.70 × 0.04) + (0.03 × 0.01) + (1.76 × 0.01) + (1.76 × 0) + (1.73 × 0.008) +(2.08 × 0.13) + (1.82 × 0.06) + (1.71 × 0.03) +(0.69 × 0.01) + (1.17 × 0.07) + (1.39 × 0.34) +(1.10 × 0.16) + (0.33 × 0.03) + 0.25 = 2.59

- Calls involving at least one IP Phone (will be needed for Gateway calculation):

  CIP = C2IP + C1IP + CSTIV + CSTID + CTSVI + CTSDI = 1632

- Calls that require DSP resources:

$$CDSP = C1IP + CT1VT + CSTID + CSTDV + CTSVD + CTSDI = 733$$

- Calls that require Virtual Trunk resources:

$$CVT = CT1VT + CT2HS + CSTIV + CSTDV + CTSVD + CTSVI = 1146$$

Real-time calculation with major applications

- ACD agent calls without Symposium = [(Number of ACD agents) × CCS/agent × 100 ÷ AHTTS] × fACD = 44 × 0.13 = 6

- Calculate the impact of other major features and applications.

  Application EBC = [(Number of application ports) × CCS per port × 100 ÷ HT] × real-time factor

  Table 74 "Application and feature EBCs and DSP requirements" (page 243) summarizes the EBC calculations. For those applications requiring DSP resources, it also provides the results of DSP calculations for applications and features, for later use.

Table 74
Application and feature EBCs and DSP requirements

Application/ Feature	Number of ports	EBC*	Required DSP ports**	Comments
Integrated Recorded Announcer	12	218	9.86	
Integrated Call Assistant	8	66	6.57	
Integrated Call Director	6	164	4.93	
CDR - outgoing		145		= 454 × 0.32
CPND		61		= 307 × 0.20, where terminating set is a TDM
Converged Desktop		398		= 171 × 2.33
Basic ACD		6		
CallPilot		2590	19.72	
Total		3648	42	

* Application EBC = (Number of application ports × CCS per port × 100 ÷ HT) × real-time factor

**Required DSP = Number of application ports × P

- Add the feature EBC to the system EBC to obtain an accurate estimate of the total CPU load:

  Total system real-time EBC = (Total system calls × real-time multiplier) + Application EBC
  = [1973 × 2.62] + 3648 = 8817

New system real-time usage Compare the total system EBC with the CPU rated capacity to determine the processor utilization.

CPU utilization = 8817 ÷ 35 000 = 25.2%

In this example, CPU utilization, including application and feature impact, is 25.2%. This loading indicates that the CPU can handle this configuration with ease and has plenty of spare capacity.

DSP calculation for Conference ports The formula to calculate the DSP requirement for conference ports is based on the number of sets in the system:

DSP channels for conference ports = (Number of TDM sets + Number of IP sets) × 0.028 × P = 550 × 0.028 × 0.82 = 13

DSP calculation without applications DSP calls (CDSP) = C1IP + CT1VT + CSTID + CSTDV + CTSVD + CTSDI = 733

DSP CCS = CDSP × WAHT ÷ 100 = 733 × 137 ÷ 100 = 1004 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 ÷ 1822 × 64 = 35

DSP and Media Card calculations Total DSP ports = DSP for calls + Conference + Applications/features = 35 + 13 + 42 = 90

Number of 32-port Media Cards required = 90 ÷ 32 = 3

For an 8-port Media Card, number of Media Cards required = 90 ÷ 8 = 12

> ***Note:*** Nortel recommends to round up the Media Card calculation to an integer.

Virtual Trunk calculation VT calls (CVT) = CT1VT + CT2HS + CSTIV + CSTDV + CTSVD + CTSVI = 1146

VT CCS = CVT × WAHT ÷ 100 = 1146 × 137 ÷ 100 = 1570 CCS

Refer to a Poisson table (with P.01 GoS) to find the corresponding number of trunks. For 1570 CCS, the number of Virtual Trunks is 61.

Number of H.323 Virtual Trunks = 61 × 0.29 = 18
Number of SIP Virtual Trunks = 61 × 0.71 = 43

User input for number of Virtual Trunks was 68. Since this is greater than the calculated number (61), the input values of H.323 Virtual Trunks = 20 and SIP Virtual Trunks = 48 should be used for further resource calculation.

Signaling Server calculation

The following information was obtained from previous calculations or input data:

> Number of IP Phones in the system = 450
> Number of Virtual Trunks = 68 (H.323 = 20; SIP = 48)
> Calls involving at least one IP set (CIP) = 1632
> Calls involving Virtual Trunks (CVT) = GKC0 = 1146

The following is additional user input to the NNEC tool:

> Endpoints served by this Gatekeeper: 100
> NRS entries (CDP + UDP + ...): 1000
> Virtual Trunks from other endpoints served by this NRS: 400
> 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 sets: Yes
> Sharing database with other traffic: Yes

PD/CL/RL database calculation (SSDB)   SSDB = b

The PD/CL/RL feature is available and sharing is allowed.

Network Routing Service calculation

> SSNR = larger of:

> {
>
> **a**   NRE ÷ NREl = 100 ÷ 5000 = 0.02   endpoints
>
> **b**   NRD ÷ NRDI = 1000 ÷ 20 000 = 0.05   dial plan entries
>
> **c**   NRC ÷ NRCHL   calls per hour
>
> }

> NRC0 is obtained from the main switch calculation.
> NRCNET = VTNET × (CCS per VT) × 100 ÷ WAHT ÷ 2
> = 400 × 26 × 100 ÷ 137 ÷ 2 = 3796
> NRC = NRC0 + NRCNET
> fH/S = (H.323 call real time) ÷ (SIP call real time) = 1800 ÷ 1200 = 1.5
> SIP calls = 48 × 26 × 100 ÷ 137 = 911
> H.323 calls = 20 × 26 × 100 ÷ 137 = 380
> NRC ÷ NRCHL = [(380 × 1.5) + 911 + 3796] ÷ 100 000 = 0.05
> This represents the loading of the Signaling Server for handling NRS calls. Compared with the results of equations (a) and (b), 0.05 is the highest of all potential usages.

Since the user wants the NRS in a dedicated Signaling Server, round up
SSNR before proceeding with further calculations:
SSNW = ROUNDUP(0.05) × 2 = 2

Terminal Proxy Server calculation

SSTR = larger of:

{

b   Since SSDB = b, with PD/CL/RL and sharing
and IPL (450) < IPLDB (1000),
IPL ÷ IPLDB = 450 ÷ 1000 = 0.45
(The database server can share the TPS function for 450 IP sets
without the need for an additional Signaling Server.)

e   IPC ÷ IPCHL = 1632 ÷ 15 000 = 0.11

}

The larger of the two values is 0.45.
Since the user wants the TPS in a dedicated Signaling Server, round up
SSTR before proceeding with further calculations:
SSTW = ROUNDUP(0.45) + 1 = 2

H.323 Gateway calculation

SSHR = larger of:

{

a   HVT ÷ HVTSL = 20 ÷ 1200 = 0.02

b   CVT ÷ HVTCHL = 380 ÷ 18 000 = 0.02

}

The larger of the two values is 0.02.
Since the user wants the H.323 Gateway in a dedicated Signaling
Server, round up SSHR before proceeding with further calculations:
SSHW = ROUNDUP(0.02) × 2 = 2

5   **SIP Gateway calculation (SSSR)**

SSSR = larger of:

{

a   SVT ÷ SVTSL = 48 ÷ 1800 = 0.03

b   CVT ÷ SVTCHL = 911 ÷ 27 000 = 0.03

}

The larger of the two values is 0.03.
Since the user wants the SIP Gateway in a dedicated Signaling Server, round up SSSR before proceeding with further calculations:
SSSW = ROUNDUP(0.03) × 2 = 2

Total Signaling Server requirement (SST)

The final calculation of SST requires picking the formula that suits the particular configuration and user input.
Since:
SSNR + SSTR + SSHR + SSSR = 0.05 + 0.45 + 0.02 + 0.03 = 0.55 < 1
SST = ROUNDUP(SSNR + SSTR + SSHR + SSSR) + 1 + (0, since SSDB = b, not c)
= ROUNDUP(0.55) + 1 = 2

The required number of Signaling Servers for this configuration is 2. The server with the database for the PD/CL/RL feature is sharing processing with all other functions within the Signaling Server.

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 = [(20 + 48) × 26] ÷ 36 = 49 erlangs

Application engineering

Contents

This section contains information on the following topics:

Introduction

Certain applications have significant capacity impact and require engineering in order to operate properly from a capacity perspective. This section provides suggestions for engineering these applications.

For descriptions of the features and their functionality, refer to feature documentation in the Nortel Technical Publications.

Converged Desktop

The Converged Desktop is a TDM or IP set configured to access Multimedia Communication Server (MCS) 5100 multimedia applications through a Session Initiation Protocol (SIP) Virtual Trunk.

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 non-SIP trunk

* calls originating from a SIP trunk but not from an MCS 5100

The PCA requirement depends only on the number of Converged Desktop users. It is independent of the percentage of voice calls using SIP trunks for conversation.

Calculating SIP access port and PCA requirements

Table 75 "SIP port and PCA requirements for Converged Desktop (with P.01 GoS)" (page 251) 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 75 "SIP port and PCA requirements for Converged Desktop (with P.01 GoS)" (page 251) 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 75
SIP port and PCA requirements for Converged Desktop (with P.01 GoS)

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
25	SIP CCS	12.5	18.1	23.8	29.4	35.0	40.6	46.2	51.9	57.5	63.1	68.8	125.0
	SIP port	3	4	4	4	5	5	5	6	6	6	7	9
	PCA	3	3	3	3	3	3	3	3	3	3	3	3
50	SIP CCS	25.0	36.2	47.5	58.8	70.0	81.2	92.5	103.8	115.0	126.2	137.5	250.0
	SIP port	4	5	6	6	7	7	8	8	9	9	10	15
	PCA	4	4	4	4	4	4	4	4	4	4	4	4

Note: Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
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

Note: Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
175	SIP CCS	87.5	126.9	166.2	205.6	245.0	284.4	323.8	363.1	402.5	441.9	481.2	875.0
	SIP port	8	9	11	13	14	16	18	19	20	22	23	37
	PCA	8	8	8	8	8	8	8	8	8	8	8	8
200	SIP CCS	100.0	145.0	190.0	235.0	280.0	325.0	370.0	415.0	460.0	505.0	550.0	1000.0
	SIP port	8	10	12	14	16	18	19	21	23	24	26	42
	PCA	8	8	8	8	8	8	8	8	8	8	8	8
225	SIP CCS	112.5	163.1	213.8	264.4	315.0	365.6	416.2	466.9	517.5	568.1	618.8	1125.0
	SIP port	9	11	13	15	17	19	21	23	25	27	28	46
	PCA	9	9	9	9	9	9	9	9	9	9	9	9
250	SIP CCS	125.0	181.2	237.5	293.8	350.0	406.2	462.5	518.8	575.0	631.2	687.5	1250.0
	SIP port	9	12	14	16	19	21	23	25	27	29	31	50
	PCA	9	9	9	9	9	9	9	9	9	9	9	9

Note: Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
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

Note: Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.

# CD Users		% voice traffic carried by SIP trunk											
		0	5	10	15	20	25	30	35	40	45	50	100
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

Note: Voice users in CCS = 5 CCS per user. Ratio of ringing time to holding time = 0.1.

D-channel

The NTBK51 Downloadable D-Channel Handler (DDCH) card mounts on the NTBK50 2.0 Mb PRI card. It provides downloadable D-channel handler interfaces based on the Multi-purpose Serial Data Link (MSDL) used in Large Systems.

Engineering considerations

The engineering guidelines assume normal traffic consisting of valid call processing and administrative messages. Engineering rules cannot prevent a piece of equipment on the network from malfunctioning and generating spurious messages, which overload the links. At this point the recovery mechanism becomes essential. The mechanism 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.

Outgoing messages originate from the system Core Processor (CP), are passed to the D-channel handler, and travel across the appropriate link to the destination. In equilibrium, or over a relatively long period of time (on the order of several minutes), the system cannot generate messages faster than the D-channel handler can process them, than the link can transmit them, or than the destination can process them. Otherwise, messages build up at the bottleneck and are eventually lost. The entity with the lowest capacity is the system bottleneck. For very short periods of time, however, one or more entities 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.

D-channel handling architecture

The D-channel handler and system exchange messages using an SRAM and interrupt scheme. To prevent any one application from tying up buffer resources, a flow-control mechanism is defined at the system and D-channel handling interface level. The flow-control mechanism is based on the common window mechanism, in which the number of messages outstanding in the transmit or receive direction per socket, or port, cannot exceed T(K) or R(K), respectively. In the transmit direction, for example, a message is considered outstanding from the time the SL-1 software writes it into the transmit ring until all processing of the message by the D-channel handler is completed. Currently T(K) and R(K) are both set at 30. Each application must queue messages if the flow control threshold is exceeded. Typically, the system task also has a buffer for messages.

An overload control threshold is also implemented in the incoming direction to protect the system Core Processor (CP) from excess messages. If the incoming messages on a single port exceed 200 messages in 2 seconds, the port is locked out, and a port overload message is printed. Manual intervention is required to clear the overloaded port. This feature prevents a single port from locking up the entire link.

Several software tasks exist on the D-channel handler. Layer 1 message processing operates at the highest priority. If the link is noisy, Layer 1 processing 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 *Software Input Output Administration (NN43001-611)*.

D-channel

For interfaces including NI-2, Q-SIG, and Euro-ISDN, Layer 3 processing is also performed on the D-channel handler, thus reducing its capacity. These interfaces are referred to as R20+ interfaces. The steady state message rate allowable for D-channel messages is 29 msg/sec for R20+ interfaces.

The SL-1 software output queue for DCH messages is the Output Buffer (OTBF), which is user-configurable for between 1 and 127 buffers in LD 17. This is a single system resource shared by all D-channels.

It is possible to define overload thresholds per D-channel for R20+ interfaces. The ISDN_MCNT (ISDN message count), defined in LD 17, specifies the number of ISDN Layer 3 call control messages allowed per

5-second interval. Overload control thresholds can be set per D-channel, ranging from 60 to 350 messages in a 5-second window, with a default of 300 messages. If the overload control threshold is exceeded, DCH421 is output. When the message rate exceeds the threshold for two consecutive 5-second periods, overload control is invoked and new incoming call requests are rejected by the Layer 3 protocol control in the third 5-second time interval. Layer 3 resumes accepting new calls at the end of the third time interval. This flexibility allows the user to regulate the processing required by a specific R20+ DCH port.

> *Note:* The default value implies no overload control, since 300 messages/5 seconds exceeds the rated capacity of 29 messages/second.

Primary Rate Interface network

Equilibrium analysis   A D-channel can be configured to support up to 383 B-channels (or 382 with a backup D-channel) on a T1 or 480 B-channels on an E1. The bandwidth available for messages is 64 kbps. Assumptions for a typical application are: 8 messages/call, 29 bytes/message, including 18 bytes of Layer 3 data and 11 bytes of Layer 2 overhead, 28 hundred call seconds (CCS)/trunk, and 180 second Average Hold Time (AHT)/call. The system capacity is derived from its call-carrying capacity for 100% incoming Primary Rate Interface (PRI) calls.

Under the traffic assumptions described above, the D-channel handler is able to support basic call processing messages for 4 D-channels under normal (steady-state) operation.

Peak analysis   When there is a link restart, STATUS messages are sent to all trunks with established calls. Since the SL-1 software task does not implement flow control on this mechanism, a burst of up to several hundred messages can be sent to the D-channel handler, exceeding flow control thresholds. When this happens, messages back up on the OTBF buffer, possibly resulting in buffer overflow, as indicated by DCH1030 messages. OTBF overflow is also possible after an initialization, since a burst of messages is sent to each D-channel in the system, and the OTBF is a shared system resource.

The system capacity is significantly higher in this scenario than in the steady state one because it is sending out D-channel messages which do not involve call processing. D-channel handling and Link capacities are also higher because, for equilibrium analysis, some capacity is reserved for peaking.

In the worst-case scenario for a single D-channel, if the system sends messages at its peak rate, OTBF buffer overflow is possible. Also, once the messages are sent, a burst of responses can be expected in the incoming direction, resulting in additional congestion at the D-channel handler.

This situation also occurs when a backup D-channel becomes active, since STATUS messages are exchanged to resynchronize the link.

To reduce the possibility of this problem occurring, limit the number of B-channels supported by a D-channel, separate D-channels onto several cards so that message bursts are not being sent to ports on the same D-channel handling card after initialization, and increase the size of OTBF to the maximum value of 127.

The Status Enquiry Message Throttle is implemented. This feature applies only to system-to-system interface networks. It 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 set in LD 17 with a range between 1 and 5. The default value is 1. Since this feature provides a flow-control mechanism for Status Enquiry messages, the likelihood of buffer overload is reduced.

B-channel overload   In an Automatic Call Distribution (ACD) environment in which the number of ACD agents plus the maximum ACD queue length is considerably less than the number of B-channels available for incoming calls, a burst of incoming messages can impact the performance of the D-channel handler as well as the system through the following mechanism: Calls from the CO terminate on a specified ACD queue. When the destination is busy (the destination set is busy or the ACD queue has reached its maximum limit of calls), the system immediately releases the call. The CO will immediately present another call to the same destination, which is released immediately by the PBX, and so on.

The B-channel Overload Control feature addresses this problem by delaying the release of an ISDN PRI call by a user-configurable time when the call encounters a busy condition. The delay in releasing the seized B-channel prevents a new call from being presented on the same B-channel, decreasing the incoming call rate. The timer BCOT is set in LD 16 with a range between 0 and 4000 msec.

ISDN Signaling Link network
In an ISDN Signaling Link (ISL) application, a modem is used to transmit ISDN signaling messages. Baud rates are user-configurable at the standard RS232/RS422 rates: 300, 1200, 2400, 4800, 9600, and 19 200 bps (see Table 76 "ISL link capacities" (page 259)). 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 Table 76 "ISL link capacities" (page 259), where the values allow for 30% peakedness.

The B-channels that can be supported assume the messaging required for a typical application as described in "Equilibrium analysis" (page 257).

Table 76
ISL link capacities

Modem baud rate	Link capacity (msgs/sec)	B-channels that can be supported
300	1 input 1 output	46
1200	4 input 4 output	180
2400	7 input 7 output	316
4800	15 input 15 output	382(T1)/480(E1)
9600	29 input 29 output	382(T1)/480(E1)
19 200	58 input 58 output	382(T1)/480(E1)

For the baud rates listed in Table 76 "ISL link capacities" (page 259), the link is the limiting constraint. The potential peak traffic problems described in "Peak analysis" (page 257) apply here as well, to an even greater extent because of the larger rate mismatch between the system and the system bottleneck. To minimize the risk, set the baud rate as high as possible.

Virtual Network Services network

Concepts applicable to ISL networks also apply to Virtual Network Services (VNS) networks. Up to 4000 VNS DNs (VDN) are supported.

D-channel bit rate The following guidelines provide the basis for engineering the Network ACD (NACD)/VNS D-channel speed.

The bit rate load on the D-channel equals:

>   the amount of messages ✕ the octets per message
>   ✕ the number of messages per second

For example, if Facility Message burst is opened with 25 calls in the queue, then the Call Request queue size is greater than or equal to 25. The outgoing facility call request is 25 messages in one second. The incoming facility call request acknowledges 25 messages in the same second. The outgoing and incoming call requests total 50 messages.

In this example, the bit rate load on the D-channel equals:

50 messages × 70 octets × 8 bits/octet
= 28 800 bits/second

Total bandwidth of a 9600 baud modem is approximately:

9600 baud × 2
= 19 200 bits/second

With a total bandwidth of 19 200 bits/second and a bit rate load of 28 800 bits/second, the D-channel cannot handle the messaging. D-channel messaging 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.

Nortel recommends that the Routing Table be designed so that Call Requests cascade to the network with the timers staggered. The node that is most likely to have available agents must 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. 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 calls 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, 100 of which are answered.

For this same network, assume now that the timers in the Routing Table are not staggered; instead, Logical Call Requests are broadcast to the 4 target nodes in the network as soon as calls arrive at the local node. Also assume that a total of 4000 calls arrive elsewhere in the network and are queued at local ACD DNs. Even if the calls are answered exactly where they were before, the number of messages exchanged increases significantly:

- 1500 calls queued on 4 ACD DNs and not answered × 4 msgs/call/DN
  = 24 000 msgs

- 500 calls answered × 11 msgs/call
  = 5500 msgs

- 500 calls queued on 3 ACD DNs and not answered × 4 msgs/call/DN
  = 6000 msgs

- 3900 network calls queued on local DN and not answered × 4 msgs/call
  = 15 600 msgs

- 100 network calls answered × 11 msgs/call
  = 1100 msgs

- Total 52 200 msgs/hr

- (52 200 msgs/hr) ÷ (3600 secs/hr) = 14.5 msgs/sec

Peak analysis When the CRQS threshold is reached, the target queue broadcasts messages to the source ACD DNs informing them that it no longer accepts 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.

If CRQS values are set high, many messages are exchanged, with the network emulating a single virtual queue. If the CRQS values are lowered, fewer Call Requests are sent across the network. However, average source delays may be increased. If FCTH levels are set too low, target nodes can bounce between Active and Closed states, resulting in network congestion and excessive real-time utilization. However, if FCTH levels are set too high, a target node may be inundated with Logical Call Request messages once it becomes available. CRQS is configurable for the range 0 to 255, while FCTH is configurable for the range 10 to 100.

Since the impact of these parameters depends on the configuration, it is not possible to make general recommendations on how to configure them. They should be determined as part of the custom network design process. Contact your local Nortel representative for network engineering services.

Impact of proper engineering of B-channels In the NACD environment, another problem arises when insufficient B-channels are configured across the network. When an agent becomes available, an Agent Free Notification message is sent to the source node. An ISDN Call Setup message is sent from the source node to the target node. Since no B-channel is available, the agent reservation timer expires, an ISDN Cancellation Message is sent from the target node to the source node, and an ISDN Cancellation Acknowledge message is sent from the source node to the target node. At this point, the agent is still free, so the process repeats until a trunk becomes available or the target closes. This scenario results in a significant amount of message passing.

Trunk requirements under Longest Idle Agent routing
Trunk requirements are usually calculated using the NACD engineering guidelines, whereby call loading for each queue at each site is estimated and used to calculate the required number of trunks between each pair of sites. However, when Longest Idle Agent (LIA) is used as the routing criterion, load estimation becomes difficult. Assuming that any agent can take any call and that agents have equal holding time characteristics, the following procedure provides a method to estimate the number of trunks required between pairs of sites.

Assumptions

1. All agents reside in one common pool and process calls at an equal rate (in other words, have a common average call service time).

2. An agent having the longest idle time occurs with equal probability among all of the agents during normal operation.

3. Agents appear as one large pool to incoming calls.

With these assumptions, under LIA, calls are routed in proportion to the number of active agents at each site.

Calculation steps

Step	Action
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 (A_i, 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) Refer to "Trunk traffic Erlang B with P.01 Grade-of-Service" (page 340) for values for Erlang B.

—End—

Parameter settings

The following are parameters that can be configured in LD 17 for CS 1000 D-channels. Items are listed with their input ranges, with default values shown in brackets.

1. OTBF 1 - (32) - 127: Size of output buffer for DCH

   This parameter configures how many output buffers are allocated for DCH messages outgoing from the system CP to the D-channel handling card. The more that are created, the deeper the buffering. For systems with extensive D-channel messaging, such as call centers using NACD, the parameter should be set at 127. For other systems with moderate levels of D-channel messaging, OTBF should be set at the smaller of the following two quantities: Total B-channels – (30 × MSDL cards with D-channels) or 127.

   For example, if a system in a standard office environment is configured with 7 T1 spans, 2 D-channels located on two different NTBK51 daughterboards, and 2 back-up D-channels, the total number of B-channels is (7 × 24) – 4 = 164. Configure OTBF to be the smaller of 164 – (30 × 2) = 104 and 127 which is 104.

2. T200 2 - (3) - 40: Maximum time for acknowledgment of frame (units of 0.5 secs)

   This timer defines how long the D-channel handler's Layer 2 LAPD waits before it retransmits a frame. It if does not receive an acknowledgment from the far end for a given frame before this timer expires, it 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 waits without receiving any frames from the far end. If no frames are received for a period of T203 seconds, the Layer 2 sends a frame to the other side to check that the far end is still alive. The expiration of this timer causes the periodic "RR" or Receiver Ready to be sent across an idle link. Setting this value too low causes unnecessary traffic on an idle link. However, setting the value too high 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 resends a frame if it does not receive an acknowledgment from the far end. Every time a

frame is sent by Layer 2, it expects to receive an acknowledgment. If it does not receive the acknowledgment, it 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 system is connected to a system that does not support the 260-byte I-frame.

6. K 1 - (7): Maximum number of outstanding frames

    This value defines the window size used by the Layer 2 state machine. The default value of 7 means that the Layer 2 state machine sends up to 7 frames out to the link before it stops and requires an acknowledgment for at least one of the frames. A larger window allows for more efficient transmission. Ideally, the Layer 2 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 D-channel handler processing required by a specific R20+ DCH port. The default value of 300 messages/5 seconds is equivalent to allowing a single port to utilize the full real-time capacity of the D-channel handler. To limit the real-time utilization of a single R20+ DCH port to $(1 \div n)$ of the real-time capacity of the D-channel handler, for $n > 1$, set ISDN_MCNT to $(300 \div n) \times 1.2$, where the 1.2 factor accounts for the fact that peak periods on different ports are unlikely to occur simultaneously. For example, to limit a single port to one-third of the processing capacity of the D-channel handler, ISDN_MCNT is set to $(300 \div 3) \times 1.2 = 120$.

If the ISDN_MCNT threshold is exceeded for one 5-second period, error message DCH421 is printed. If the threshold is exceeded for two consecutive periods, incoming call requests arriving in the third 5-second interval are rejected by the D-channel handler Layer 3 software. At the end of the third 5-second interval, Layer 3 resumes accepting incoming call requests.

Serial Data Interface (SDI)

The SDI network interface on the Small System Controller (SSC) cards in the Media Gateways and on the Terminal Server provide an asynchronous serial data 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.

Normally, in the output direction, the SDI Application passes any character received from the system to the Layer 1 Driver to be sent out over the interface. If XON/XOFF Handling is enabled for printing, the SDI Application buffers up to 500 characters once an XOFF is received. The system is not aware that an XOFF has been received. After the buffer is full, if further output is received, the oldest data is discarded. Output resumes when an XON is received or one minute has passed since the output was halted by an XOFF. At this point, the contents in the buffer are emptied first, followed by output from the system. If any data has been discarded, an error message is sent.

In the input direction, every character received by the Layer 1 Driver 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 bounce between a modem or printer and the system. To avoid these situations, configure modems in dumb mode and disable printer flow control.

The system input buffer is the TTY input buffer, which can store 512 characters. The system output buffer is the TTY output buffer, which can store 2048 characters.

Call Detail Records

Call Detail Recording (CDR) records are available in two formats: *FCDR=old* and *FCDR=new*. A typical record for the old format is 100 bytes long while a typical record for the new format is 213 bytes long (see Table 77 "Link capacities for CDR application (outgoing)" (page 267)). 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 77
Link capacities for CDR application (outgoing)

Modem baud rate	Link capacity (msg/sec) (peak)	Calls/Hour for *FCDR=old*	Call/Hour for *FCDR=new*
300	30	831	390
1200	120	3323	1560
2400	240	6646	3120
4800	480	13 292	6241
9600	960	26 585	12 481
19 200	1920	53 169	24 962
38 400	3840	106 338	49 924

Equilibrium analysis   The system capacity for messages per second is conservatively based on the assumption of 100% outgoing calls with *FCDR=new*. Typically, CDR records are not generated for 100% of the calls.

Peak analysis   Since each character is sent as a separate message, every time a CDR record is sent, a traffic peak is generated.

To prevent system buffers from building up, set the baud rate at 38 400. If a lower baud rate is chosen, assume that the CDR application 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.

D-channel handler engineering procedure

It is important to engineer the D-channel handler in the context of engineering the entire system. Refer to *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for additional information on real-time engineering of the system. In all cases with a user-configurable link rate, it is essential that the link be configured so that the rate is high enough to support steady-state requirements and some peakedness. Otherwise, the application messages will occupy system buffers, increasing the chance of buffer overflow.

Table 78 "D-channel handler engineering worksheet" (page 268) is a high-level worksheet for analysis of D-channel handling capacity. See Table 84 "Real-time requirements for D-channel applications" (page 274) through Table 82 "Peak buffer requirements for SDI applications" (page 271) for the values to use in the worksheet.

Table 78
D-channel handler engineering worksheet

Port	Application	Real Time required	Peak Buffer usage outgoing	Peak Buffer usage incoming
0	_____	_____	_____	_____
1	_____	_____	_____	_____
2	_____	_____	_____	_____
3	_____	_____	_____	_____
Total		_____	_____	_____

Assuming 30% peakedness for the applications, the total real time required should be less than 2 770 000 msec. The projected real-time utilization of the D-channel handler is given by:

Real-time usage = Total Real Time Required ÷ 2 770 000

Nortel recommends that peak buffer usage be less than 60 in each direction. As the peak buffer usage increases over 60, the likelihood of an intermittent buffer full problem increases.

The following sections provide procedures, including worksheet tables, for calculating the real time required on the D-channel handler for various applications.

In Table 78 "D-channel handler engineering worksheet" (page 268) through Table 82 "Peak buffer requirements for SDI applications" (page 271), 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 78 "D-channel handler engineering worksheet" (page 268), and the appropriate Peak Buffer Usage values should be inserted in the corresponding Peak Buffer Usage columns of Table 78 "D-channel handler engineering worksheet" (page 268).

DCH applications

If several applications share a D-channel, add the final real-time requirements for the applications and then enter the total in the appropriate entry in .

Table 79
Real-time requirements for D-channel applications

DCH	Calls/hr A	Msgs/call B	Msgs/hr C = A × B	Msec/msg D	Msec E = C × D
ISDN Network	trunks/DCH × calls/hr/trunk (15.6) = _____ 8		_____	pre-R20: 8.8 R20+: 26.5	_____
NACD	NACD agents × calls/hr/agent (18.3) = _____ 30		_____	pre-R20: 8.8	_____
NMS	NMS ports (see note) × calls/hr/port (65) = _____ 10		_____	pre_R20: 8.8	_____
Note: For clarification of the terms "pre-R20" and "R20+," refer to "D-channel" (page 256)					

The calculations described for NACD provide a simplified approximation of a "typical" NACD network. If call flows can be predicted or estimated, they can be used to develop a more accurate model using the number of messages. When this is done, the msgs/hr is computed directly, so columns A and B are not used. See "Examples" (page 272) for a detailed example of how this can be done.

If a live system is being modeled, add the "number of all incoming messages received on the D-channel" and the "number of all outgoing messages sent on the D-channel" field from a busy hour TFS009 report to derive the entry for Column C. See *Traffic Measurement Formats and Outputs Reference (NN43001-750)* for details.

Table 80 "Peak buffer requirements for D-channel applications" (page 270) shows peak buffer requirements for D-channel applications.

Table 80
Peak buffer requirements for D-channel applications

DCH	Outgoing	Incoming
ISDN Network	SEMT (1) × 8	SEMT (1) × 8
NACD	Source ACD DNs + 5 = _____	Network congestion level: • Low: 10 • Medium: 20 • High: 30
NMS	10	10

In the case of an ISL D-channel, ensure that the baud rate of the connection is greater than

$$(C \text{ msgs/hr} \times 29 \text{ bytes/msg} \times 8 \text{ bits/byte}) \div 3600 \text{ sec/hr}$$

where C comes from column C in Table 84 "Real-time requirements for D-channel applications" (page 274).

If the baud rate is too low to meet requirements, performance of the entire D-channel handler may be jeopardized since 30 of the 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.

SDI applications

In the HSL analysis, include live agents, automated agents, and CallPilot agents in the agent total. This will compensate for the assumption of simple calls. Table 81 "Real-time requirements for SDI applications" (page 271) shows real-time requirements for SDI applications.

Table 81
Real-time requirements for SDI applications

SDI	calls/hr A	msgs/call B	msgs/hr C=A×B	msec/msg D	msec E=C ×D
CDR	calls/hr with reports = _____	FCDR = old:100 FCDR = new: 213	_____	0.05	_____
HSL	agents × calls/agent/hr (18.3) = _____	5			
			_____	8.8	_____
TTY	NA	NA	15 000	0.05	_____

There are no traffic reports that provide information on the number of SDI messages directly. For CDR records, determine whether CDR is enabled for incoming, outgoing, and/or internal calls. The number of incoming, outgoing, internal, and tandem calls is available from TFC001. Tandem calls are considered both incoming and outgoing. Alternatively, the number of CDR records can be counted directly.

Table 82 "Peak buffer requirements for SDI applications" (page 271) shows peak buffer requirements for SDI applications.

Table 82
Peak buffer requirements for SDI applications

SDI	Outgoing	Incoming	Minimum baud rate
CDR	• 30 if baud rate is less than recommended in Table 77 "Link capacities for CDR application (outgoing)" (page 267)	1	(msgs/hr × 10 bits/msg) ÷ (3600 sec/hr) = ____

SDI	Outgoing	Incoming	Minimum baud rate
HSL	• Messages per call — simple: 5 — medium: 10 — complex: 15	1	(msgs/hr × 20 bytes/msg × 9 bits/byte) ÷ (3600 sec/hr) = ____
TTY	10	10	

Examples

NACD network with CDR reports

Consider an NACD network with the topology given in . The call flow is provided, where arrows indicate where calls enter the network and where they are answered.

Each node has a single ACD DN and calls are queued to the network target DNs as soon as they arrive.

For this network, determine whether a single D-channel handler on Node B can support DCH1, DCH2, and an SDI port for CDR records on Port 0.

With the detailed call-flow information, a messaging model for DCH1 and DCH2 can be developed (see).

Figure 42
NACD network

Table 83
NACD Message Model

Originating Node	Total Queued	Queued and answered	Queued but not answered	Total messages	DCH1	DCH2
Node A to Node B	3000	500	2500	15 500	x	x
Node A to Node C	3000	500	2500	15 500	x	x
Node B to Node A	2600	100	2500	11 100	x	
Node B to Node C	2600	500	2100	13 900		x
Node C to Node A	1650	50	1600	6950	x	x
Node C to Node B	1650	100	1550	7300	x	x

The DCH1 and DCH2 columns indicate whether the messages 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, the total messages are derived 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 84 "Real-time requirements for D-channel applications" (page 274)).

Table 84
Real-time requirements for D-channel applications

DCH	calls/hr A	msgs/call B	msgs/hr C=A×B	msec/msg D	msec E=C×D
NACD DCH1	NA	NA	56 350	pre-R20: 8.8	495 880
NACD DCH2	NA	NA	59 150	pre-R20: 8.8	520 520

Assuming that no non-NACD calls are carried, Node B carries 3750 calls/hour.

Table 85
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, as shown in Table 86 "Engineering worksheet" (page 274).

Table 86
Engineering worksheet

Port	Application	Real Time required	Peak Buffer usage outgoing	Peak Buffer usage incoming
0	CDR	39 938	10	1
1	DCH-NACD	495 880	7	10
2	DCH-NACD	520 520	7	10
3				
Total		1 056 338	24	21

The projected D-channel handler utilization is 1 056 338 ÷ 2 770 000 = 38%. Assuming low network congestion, incoming and outgoing peak buffer usage are below 60, so a single D-channel handler is able to support this configuration. However, due to the potentially high messaging impact of NACD, this should be re-engineered periodically to determine whether the call volumes or call flow patterns have changed.

Meridian Mail
Traffic calculations and capacity table

Refer to *Site and Installation Planning (553-7011-200)* for a detailed engineering of Meridian Mail (MM), including menu utilization, call duration, storage size, disk size, up requirements, and so on. However, for easy reference, a simplified table is extracted and included here (see Table 87 "Meridian Mail channel capacity" (page 275)).

Each Meridian Mail Module consists of 16 ports that interface with a DTI type of loop with 24 ports to provide voice channels. In other words, every 16 Meridian Mail ports interface with one loop of 30 timeslots.

As with other traffic calculations, the first step is to determine the average holding time of an MM call. This includes both the time the user is logged on to MM and the time callers are leaving messages for that user. A typical range is 30 to 60 seconds per user depending on the type of application.

The calling rate per MM registered user is approximately 10% of busy hour calls. For example, if a set generates or receives five calls per hour, the MM calls would be 0.5 per hour. If there are 2000 MM users in a switch with average holding time (AHT) of 60 seconds, its MM traffic would be:

MM traffic in CCS = 2000 × 0.5 × 60 ÷ 100 = 600 CCS

From Table 87 "Meridian Mail channel capacity" (page 275), approximately 23 MM ports are needed for this application.

Note that if complicated voice menus are involved for an application, the AHT needs to reflect that fact.

Table 87
Meridian Mail channel capacity

Number of channels	Capacity in CCS
4	54
8	157
12	273
20	522
24	651
28	782
32	915
36	1049
40	1183
44	1318
48	1455

Number of channels	Capacity in CCS
52	1592
56	1729
60	1867
64	2005
68	2143
72	2282
76	2421
80	2561
84	2700
88	2840
92	2980
96	3120

The main objective of presenting the Meridian Mail engineering procedure here is to show how it fits into the overall Call Center engineering in "CallPilot engineering" (page 276). For a high-level MM port requirements estimate, interpolation or extrapolation between entries is permitted.

The major MM parameter that impacts the real-time capacity of a collocated system is the type of signaling between the MM processor and the system CP. For locally generated MM calls, CSL and End to End signaling have significant capacity effects and have different real-time factors as shown in the real-time calculation worksheet.

Many voice processing features are offered with the Meridian Mail application, all of which present unique characteristics in MM usage. Each specific feature, with varying AHT, impacts the MM port requirement differently. This must be considered when engineering a specific MM application. The following are known applications of the MM feature: Voice Mail, Voice Menu, Voice Forms, Auto Attendant, Meridian Interactive Voice Response (MIVR), Host Enhanced Voice Processing (HEVP), Network Message Service, and Third Party Voice Messaging Systems.

CallPilot engineering

Refer to *CallPilot Planning and Engineering (*553-7101-101) for engineering details. The abbreviated procedure in this chapter is for system engineering where a rough estimate of CallPilot ports (or channels) is required.

In addition to voice channels, a CallPilot allows fax and speech-recognition media. As a measure of Digital Signal Processing (DSP) power, different media types require different Multimedia Processing Unit (MPU) quantities:

- One voice channel requires one MPU

- One fax channel requires two MPUs

- One speech-recognition channel requires four MPUs

A Multimedia Processing Card (MPC-8) is a credit-card sized PC card that resides in the CallPilot Server. Each MPC-8 has eight MPUs. The maximum number of MPUs in a CallPilot is 96. Any use of non-voice application reduces the number of channels available for voice traffic.

For an IP source to access CallPilot, the codec must be set for G.711. Since a non-standard proprietary codec is used in CallPilot, a multi-rate transcoding 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. Refer to *CallPilot Planning and Engineering (*553-7101-101) for details.

Once the CCS for each type of media is calculated, calculate the total and refer to capacity tables in the NTP for the MPU requirement based on the offered CCS traffic.

Call Center

The Call Center is an ACD switch whose calls are mostly incoming or outgoing and with extensive applications features, such as Nortel Hospitality Integrated Voice Services. A port in the Call Center environment, either as an agent set or trunk, tends to be more heavily loaded than other types of applications.

System capacity requirements depend on customer application requirements, such as calls processed in a busy hour, and feature suites such as Recorded Announcement (RAN), Music, and Interactive Voice Response (IVR).

ACD

Automatic Call Distribution (ACD) is an optional feature available with the system. It is used by organizations where the calls received are for a service rather than a specific person.

For basic ACD, incoming calls are handled on a first-come, first-served basis and are distributed among the available agents. The agent that has been idle the longest is presented with the first call. This ensures an equitable distribution of incoming calls among agents.

The system is managed or supervised by supervisors who have access to the ACD information through a video display terminal. These supervisors deal with agent-customer transactions and the distribution of incoming calls among agents.

Many sophisticated control mechanisms have been built on the basic ACD features. Various packages of ACD features 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 network interfaces for communications. Call processing and service management functions are split between the system and the auxiliary processor.

ACD-MAX

ACD-MAX offers a customer managerial control over the ACD operation by providing past performance reporting and current performance displays. It is connected through an SDI port to communicate with the system CP. The ACD-MAX feature makes the necessary calculations of data received from the system to produce ACD report data for current and past performance reports. Every 30 seconds, ACD-MAX takes the last 10 minutes of performance data and uses it to generate statistics for the current performance displays. The accumulated past performance report data is stored on disk every 30 minutes.

ACD-MAX calls impact capacity engineering in the real-time area only.

NACD

The majority of tasks in the engineering of Network ACD (NACD) involve the design of an NACD routing table and the engineering of overflow traffic. The complexity of the process is beyond the scope of this document. The engineering procedure in this NTP is for single-node capacity engineering, which accounts for the real-time impact of NACD calls on a switch either as a source node or remote target node. Therefore, the overall design of a network is not in the scope of this document.

RAN and Music

The RAN trunk can be treated just like a normal trunk. The only potential capacity impact is for systems that include RAN trunks in blocking or non-blocking calculations. The calculations determine the total number of loops or card slots required.

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

Refer to "Service loops and circuits" (page 176) for more information.

Symposium Call Center

Symposium is a Host Server that interfaces through an Ethernet to enable the system to provide advanced Call Center features to users. Although Internet Protocol (IP) is used for communications, the underlying message to the system input queue is an Application Module Link (AML) message.

The customer can create simple-to-write scripts in Symposium to control processing of an arriving call that is eventually delivered to an agent queue after following various call processing rules, such as skill set of agent, call priority, and length of waiting time.

The complexity of call handling on the system call processor determines the impact of Symposium Call Center on the system. Depending on the script used, the call processing can include giving RAN, Music, and IVR, all of which require a voice-processing system such as CallPilot.

Symposium Call Center with IP phones and Virtual Trunks

When IP phones are used as ACD agent sets, there are certain special engineering rules. Two additional resources must be engineered:

- Digital Signal Processor (DSP) channels (therefore, Media Cards)

- Virtual Trunks

Refer to "Resource calculations" (page 205) for the detailed calculations.

ELAN engineering

The Embedded Local Area Network (ELAN) subnet is designed to handle messaging traffic between the system and its applications, such as Symposium and CallPilot. It is not designed to handle functions 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 with data rate of 10 Mbps or 100 Mbps, will not be a bottleneck in a Symposium/CallPilot configuration. However, observe the following engineering guidelines to avoid performance problems. For more detailed information, refer to *Converging the Data Network with VoIP Fundamentals (NN43001-260)*.

- Ensure that settings on the physical interface of the system to the Ethernet are correct.

- Although no traffic engineering is required on the ELAN subnet, if the loading on link is extremely high (for example, above 10% on the 10T-10 Mbps), collision on the Ethernet is possible. Use a sniffer to detect any performance problems. Decrease the loading on the link if it is overloaded.

- Set a consistent data rate with the application.

Certain remote maintenance applications can utilize the ELAN subnet to access the system from a remote location. Ensure that no other enterprise IP network traffic is introduced.

CLASS network engineering rules

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 non-blocking network is achievable.

Feature operation

A call originated from Set A (or Trunk A) seeks to terminate on a Custom Local Area Signaling Services (CLASS) Set B. When B starts to ring, A will hear 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 set.

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 Set B, A's CND is delivered to CMOD through SSD messages (using signaling channel only). When the CND information is sent from CMOD to CLASS Set B, it is delivered through a voice path during the four-second silence cycle of Set B. The CMOD unit is held for a duration of six seconds.

The system delivers SSD messages containing CND information to CMOD and then sends it to Set B during the delivery interval through a voice path.

Table 88 "CMOD Unit Capacity" (page 281) is the CMOD capacity table. It provides the number of CMOD units required to serve a given number of CLASS sets with the desired GoS (P.001). The required number of CMOD units should have a capacity range whose upper limit is greater than the number of CLASS sets equipped in a given configuration

Table 88
CMOD Unit Capacity

CMOD Unit	CLASS Set	CMOD Unit	CLASS Set
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 Set	CMOD Unit	CLASS Set
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 non-Call Center applications

In a non-call center application, there is no significant number of agent sets. Therefore, no agent set to regular set conversion is needed.

Engineering rule (no reconfiguration required)

The following engineering rule should be followed to avoid the need to reconfigure a switch to accommodate the CLASS feature: Provide the number of CMOD units serving all CLASS sets in the system based on the capacity table (see Table 88 "CMOD Unit Capacity" (page 281) above).

Guidelines for Call Center applications
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 sets to regular sets:

   1 agent CLASS set = 4 sets (called equivalent sets)

2. Calculate the total number of regular CLASS sets and equivalent CLASS sets and find the number of CMOD units required based on the capacity table (see Table 88 "CMOD Unit Capacity" (page 281)).

Configuration parameters

Design parameters are constraints on the system established by design decisions and enforced by software checks. Defaults are provided in the factory-installed database. However, some parameter values must be set manually, through the OA&M interface, to reflect the actual needs of the customer's application.

For guidelines on how to determine appropriate parameter values for call registers, I/O buffers, and so on, see "Design parameters" (page 143) and "Memory engineering" (page 154).

Provisioning

Contents

This section contains information on the following topics:

Introduction

This chapter outlines the procedures required to determine equipment requirements.

Provisioning a new system

Perform the following tasks to provision a new system:

1. Define and forecast growth ("Defining and forecasting growth" (page 286)).

2. Estimate CCS per terminal ("Estimating CCS per terminal" (page 287)).

3. Calculate number of trunks required ("Calculating number of trunks required" (page 291)).

4. Calculate line, trunk, and console load ("Calculating line, trunk, and console load" (page 292)).

5. Calculate DTR requirements ("Calculating Digitone receiver requirements" (page 292)).

6. Calculate CS 1000M resources ("Calculating CS 1000M resources" (page 295))

7. Calculate total system load ("Calculating total system load" (page 295)).

8. Calculate number of loops required ("Calculating number of loops required" (page 295)).

9. Calculate number of IPE cards required ("Calculating number of IPE cards required" (page 295)).

10. Provision Conference/TDS loops ("Provisioning conference/TDS loops" (page 296)).

11. Calculate memory requirements ("Calculating memory requirements" (page 297)).

12. Assign equipment and prepare equipment summary ("Assigning equipment and preparing equipment summary" (page 297)).

13. Calculate battery backup time ("Calculating battery backup time" (page 297)).

Defining and forecasting 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 180 employees and needs 100 telephones to meet the system cutover. The customer projects an annual increase of 5% of employees based in future business expansion. The employee growth forecast is:

- 180 employees x 0.05 (percent growth) = 9

- 189 employees x 0.05 = 10 additional employees at 1 year

- 199 employees x 0.05 = 10 additional employees at 2 years

- 209 employees x 0.05 = 10 additional employees at 3 years

- 219 employees x 0.05 = 11 additional employees at 4 years

- 230 employees x 0.05 = 12 additional employees at 5 years

The ratio of telephones to employees is 100/180, which equals 0.556.

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 employees (0.556).

- 180 employees x 0.556 = 100 telephones at cutover

- 189 employees x 0.556 = 105 telephones at 1 year

- 199 employees x 0.556 = 111 telephones at 2 years

- 209 employees x 0.556 = 116 telephones at 3 years

- 219 employees x 0.556 = 122 telephones at 4 years

- 230 employees x 0.556 = 128 telephones at 5 years

This customer requires 100 telephones at cutover, 111 telephones at two years, and 128 telephones at five years.

Each DN assigned to an analog (500/2500-type) telephone requires a TN. Determine the number of analog (500/2500-type) TNs required for each customer and enter this information in "Worksheet "Worksheet 1: Growth forecast" (page 302). Perform this calculation for cutover, two-year and five-year intervals.

Estimating CCS per terminal

Estimate the station and trunk CCS per terminal (CCS/T) for the installation of a system using any one of the following methods:

- comparative method

- manual calculation

- default method

Nortel Communication Server 1000
Communication Server 1000M and Meridian 1 Small System Planning and Engineering
NN43011-220   01.01   Standard
Release 5.0   30 May 2007

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Comparative method

Select three existing systems which have a record of traffic study data. The criteria for choosing comparative systems are:

- similar line size (+ 25%)

- similar business (such as bank, hospital, insurance, manufacturing)

- similar locality (urban or rural)

Once similar systems have been selected, their station, trunk, and intra CCS/T are averaged. The averages are then applied to calculate trunk requirements for the system being provisioned (see the example in Table 89 "Example of station, trunk, and intra CCS/T averaging" (page 288)).

Table 89
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	3.0	2.0	7.60	2.50
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 90 "Example of CCS/T averaging when only trunk CCS/T is known" (page 288)).

Table 90
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
DID	16	P.01	294	234	1.20
CO	14	P.02	267	234	1.14
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	4	20.00
			Total: 959		Total: 23.78

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) = 959/234 = 4.1

Assuming a 33% intra-calling ratio:

intra CCS/T = 4.1 * 0.5 = 2.1

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 to be maintained at two-year and five-year periods (see). (If not, use the comparative method described on .)

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 91
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 tolerates a lesser grade of service on these trunk groups. Table 92 "Estimated load per trunk" (page 290) lists the estimated usage on special services trunks.

Table 92
Estimated load per trunk

Trunk type	CCS
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 two 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 93 "Default method and manual calculations analysis" (page 291) 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 93
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

Calculating number of trunks required

Enter the values obtained through any of the three previous methods in "Worksheet . Add the calculations to the worksheet. Once the trunk CCS/T is known and a grade of service 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 grade of service (see reference table). 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 reference table 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.

Calculating line, trunk, and console load

Once the quantity of trunks required has been estimated, enter the quantities in "Worksheet "Worksheet 1: Growth forecast" (page 302) 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 in "Worksheet "Worksheet 1: Growth forecast" (page 302).

Line load

Line load is calculated by multiplying the total number of 500-telephone TNs by the line CCS/T. The number of TNs is determined as follows:

- one TN for every DN assigned to one or more Analog (500/2500-type) telephone

- one TN for every Meridian Digital Telephone without data option

- two TNs for every Meridian Digital Telephone with data option

Trunk load

Trunk load is calculated by multiplying the total number of digital telephone and 500-line TNs which 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.

Calculating Digitone receiver requirements

The NTDK20 SSC card meets all DTR requirements. DTR provisioning methods are provided below for exceptional cases requiring extra DTR capacity.

The Cabinet system has 50 universal card slots when four Expansion Cabinets are equipped. The maximum possible number of lines is therefore:

50 cards * 16 units/card = 800 lines

"Digitone receiver requirements Model 1" (page 344) through "Digitone receiver requirements Model 4" (page 346) are based on models of traffic environments and can be used 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. The number of DTRs must accommodate the highest result.

Some special features are:

- Authorization Code

- Centralized Attendant Service (CAS)

- Charge Account for Call Detail Recording (CDR)

- Direct Inward System Access (DISA)

- Integrated Messaging System Link

From the appropriate reference table ("Digitone receiver requirements Model 1" (page 344) through "Digitone receiver requirements Model 4" (page 346)), determine the number of DTRs required and the DTR load for cutover, two-year, and five-year intervals. Record this information in "Worksheet "Worksheet 2: Total load" (page 303).

The following models are based on some common PBX traffic measurements.

Model 1

"Digitone receiver requirements Model 1" (page 344) is based on the following factors:

- 33% intra-office calls, 33% incoming calls, and 33% outgoing calls

- 1.5% dial tone delay grade of service

- no Digitone DID trunks or incoming Digitone TIE trunks

Model 2

"Digitone receiver requirements Model 2" (page 344) is based on the following factors:

- the same traffic pattern as Model 1

- Digitone DID trunks or incoming Digitone TIE trunks

- Poisson 0.1% blocking grade of service

Model 3

"Digitone receiver requirements Model 3" (page 345) is based on the following factors:

- 15% intra-office calls, 28% incoming calls, and 56% outgoing calls

- 1.5% dial tone delay grade of service

- no Digitone DID trunks or incoming Digitone TIE trunks

Model 4

"Digitone receiver requirements Model 4" (page 346) is based on the following factors:

- the same traffic pattern as Model 3

- Digitone DID trunks or incoming Digitone TIE trunks

- Poisson 0.1% blocking Grade-of-Service (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 intra-office and outgoing call calculations is 135 seconds if unknown.

- Digitone receiver holding times are 6.2 and 14.1 seconds for intra and outgoing calls respectively.

- Factor (1 - R)/ 2 in (1) outgoing (incoming calls and outgoing calls are equal). R is the intra-office ratio.

1. Follow the procedure below for detailed calculation Method 1.

    a. Calculate Digitone calls:

    $$\text{Intra-office traffic} = \frac{100 * \text{Digitone station traffic (CCS)}}{\text{call holding time in seconds}} * \frac{R}{2}$$

    $$\text{Outgoing traffic} = \frac{100 * \text{Digitone station traffic}}{\text{call holding time in seconds}} * \frac{(1-R)}{2}$$

    b. Calculate total DTR traffic:

    $$\text{Total DTR traffic} = \frac{1.3 * [(6.2 * \text{intra}) + (14.1 * \text{outgoing})]}{100}$$

    c. Calculate average holding time:

    $$\text{Average holding time} = \frac{(6.2 * \text{intra}) + (14.1 * \text{outgoing})}{(\text{intra calls} + \text{outgoing calls})}$$

2. See and use the answers from steps 1(b) and (c) 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 the procedure below for detailed calculation Method 2:

1. Calculate intra-office and outgoing Digitone calls as shown in step 1 of Method 1:

$$DID \ calls = \frac{100 * Digitone \ station \ traffic \ (in \ CCS)}{call \ holding \ time \ in \ seconds}$$

2. Calculate total DTR traffic:

$$DTR \ traffic = \frac{1.3 * [(6.2 * intra) + (14.1 * outgoing)] + (2.5 * DID \ calls)}{100}$$

3. See "Digitone receiver requirements Poisson 0.1 percent blocking" (page 350) and use the answer from step 2 to determine the number of DTRs required.

Calculating CS 1000M resources

A CS 1000M Small System requires the following additional resources:

- Digital Signal Processor (DSP) resources provided by Voice Gateway Media Cards

- Session Initiation Protocol (SIP) and H.323 IP Peer Virtual Trunks

- One or more Signaling Servers

Use "Worksheet "Worksheet 11: Media Card calculation" (page 316) and "Worksheet "Worksheet 12: Signaling Server calculation" (page 318) to calculate the required resources. Refer to "Resource calculations" (page 205) for more information.

Calculating 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 into "Worksheet "Worksheet 2: Total load" (page 303).

Calculating number of loops required

Loop provisioning is not required with Small Systems since each card is automatically assigned to its own loop. By default, the system is non-blocking.

Each cabinet can house up to ten Intelligent Peripheral Equipment (IPE) cards.

Each chassis can house up to four IPE cards.

Calculating number of IPE cards required

Using information from "Worksheet "Worksheet 1: Growth forecast" (page 302), enter the number of Meridian Digital Telephone TNs, analog (500/2500-type) TNs, and trunk TNs required at cutover, two-year, and five-year intervals (for all customers) in "Worksheet "Worksheet 3: System cabinet/chassis requirements" (page 304).

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.

Refer to "Card slot assignments for the Cabinet system" (page 129) and "Card slot assignments for the Chassis system" (page 135) for information on how to allocate the cards.

Provisioning conference/TDS loops

Conference loops

The Conference function is provided by the NTDK20 SSC card.

Each conference loop supports 16 conferees. By default, two conference loops are always active. More become active when the Expansion Cabinets or Chassis are equipped. Therefore, the SSC supports a total of 32 conferees by itself.

Each port on a Fiber or IP Expansion daughterboard on the SSC card supports an additional conference loop for a total of:

- 48 conferees when equipped with one Expansion Link

- 64 conferees when equipped with two Expansion Links

- 80 conferees when equipped with three Expansion Links

- 96 conferees when equipped with four Expansion Links

TDS loops

The SSC card has 60 channels of TDS. This should be enough to meet all TDS requirements.

To illustrate this point, the following two examples are provided.

Example 1

A Cabinet system configured with two Expansion Cabinets provides 30 slots for trunk and line cards.

The SSC card can support 7260 CCS of call traffic. A digital line card supports 16 units per card. A Universal trunk card supports 8 units per card.

The CCS per card would be:

| Digital line card | 16 units/card x 6 CCS/unit = 96 CCS/card |
| Universal trunk card | 8 units/card x 22 CCS/unit = 176 CCS/card |

Assume the following:

- An average station generates 6 CCS of traffic.

- A 20% trunking ratio.

- An average trunk generates 22 CCS of traffic.

The 50 card slots available can support a system configuration of 640 lines (40 line cards) and 80 trunks (10 trunk cards). The total CCS for this configuration will be:

Total CCS: (40 line cards x 96 CCS/card) + (10 trunk cards x 176 CCS/card)

= 3840 CCS + 1760 CCS

= 5600 CCS

This number is less than the TDS capability of the SSC card. If the calculation yields a number greater than one, you can add an XDTR card to the system.

Example 2
A system that is more heavily trunked, as in the case of a one-to-one ratio, can support a configuration of 320 lines (20 line cards) and 240 trunks (30 trunk cards):

Total CCS: 20 line cards x 96 CCS/card + 30 trunk cards x 176 CCS/card

= 1920 CCS + 5280 CCS

= 7200 CCS

The SSC card, at 7260 CCS, still provides sufficient TDS capability.

Calculating memory requirements
Use "Worksheet "Worksheet 5: Unprotected memory calculations" (page 307) and "Worksheet "Worksheet 4: Protected memory calculations" (page 306) to calculate memory requirements. Use the two-year figure for telephones, consoles, and trunks for the calculation. Add 10% to the total memory requirements.

Assigning equipment and preparing equipment summary
Use "Worksheet "Worksheet 6: Equipment summary" (page 308) to record the equipment requirements for the complete system at cutover. Assign the equipment. The equipment summary may require updating as a result of assignment procedures. Use the finalized equipment summary to order the equipment for the system.

Calculating battery backup time
Use this procedure to determine:

- system power consumption

- battery current for customer-provided DC reserve power

- battery backup time for the NTAK75

- battery backup time for the NTAK76

Use the circuit-card power-consumption table (Table 13 "Circuit card and daughterboard power consumption" (page 56)) and the Cabinet system power consumption and battery current calculation worksheets (see Appendix "Worksheets" (page 301)).

Procedure 1
Calculating battery backup time

Step	Action

1   Determine the circuit card configuration in each system cabinet. Record the card codes against their cabinet slot numbers, on "Worksheet "Worksheet 7: Cabinet system power consumption" (page 309).

2   For each circuit card, transfer the power consumption values from Table 13 "Circuit card and daughterboard power consumption" (page 56) to the power-consumption column on the corresponding worksheet.

3   Calculate the total system power consumption on "Worksheet 7f: Total Cabinet system power consumption" (page 311)".

4   If your system is AC-powered, go to "Worksheet "Worksheet 9: Battery current and AC line calculation for AC systems using NTAK75 and NTAK76" (page 313). If your system is DC-powered, go to "Worksheet "Worksheet 10: Battery current calculation for customer- provided DC reserve power" (page 314).

5   Transfer the P*out* (Main) and P*out* (Expan.) values from Worksheet "Worksheet 7: Cabinet system power consumption" (page 309)f to Worksheet "Worksheet 9: Battery current and AC line calculation for AC systems using NTAK75 and NTAK76" (page 313) or "Worksheet 10: Battery current calculation for customer- provided DC reserve power" (page 314).

6   Calculate P*in* (Main), I *Batt* (Main), P*in* (Expan), and I *Batt* (Expan) as shown on Worksheet "Worksheet 9: Battery current and AC line calculation for AC systems using NTAK75 and NTAK76" (page 313) or "Worksheet 10: Battery current calculation for customer- provided DC reserve power" (page 314).

7   Calculate I*line* if required, as shown on Worksheet "Worksheet 9: Battery current and AC line calculation for AC systems using NTAK75 and NTAK76" (page 313).

8   Transfer the values calculated for I Batt (Main) and I Batt (Expan), onto the NTAK75/QBL24A1 and the NTAK76 discharge time graphs (Figure 43 "Discharge time for the NTAK76 Battery" (page 315)).

9   Select the battery unit that provides the most appropriate backup time.

—End—

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Appendix A
Worksheets

List of worksheets

Introduction

The worksheets in this appendix give examples of information needed to calculate power consumption, allocate circuit cards, and do traffic and equipment engineering. However, more detailed information is needed

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to fully engineer a system. Consult your Nortel representative and use a configuration tool, such as Nortel Enterprise Configurator (NNEC), 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" (page 205). The result of the worksheet is a number or set of numbers, in the units of the capacity being assessed.

Worksheet 1: Growth forecast

Customer:_____

Date: _____

Prepare one worksheet for each customer and one worksheet for the complete system.

Stations	Cutover	2 years	5 years	CCS/T
Meridian Digital Telephones				
Meridian Digital Telephone TNs				
500 telephones				
500 TNs				
2500 telephones				
2500 TNs				
2-way				
1-way in				
1-way out				
DID				
TIE				
CCSA				
InWATS				
OutWATS				
FX				
Private line				
Dial dictation				
Paging				

Stations	Cutover	2 years	5 years	CCS/T
RAN				
AIOD (Automatic ID of Outward Dialing)				
DTI				
E&M 2W				
E&M 4W				
CO				

Line CCS/T_____

Total trunk CCS/T_____

Intra CCS/T_____

Worksheet 2: Total load

Customer:_____

Date: _____

Prepare one worksheet for each customer for cutover, 2-year, and 5-year intervals, and one worksheet for the system for cutover, 2-year, and 5-year intervals.

Line usage

Meridian Digital sets: TN_____ x _____CCS/T=_____ CCS

500: TN_____ x _____ CCS/T=_____ CCS

2500: TN_____ x _____ CCS/T=_____ CCS

Total line load=_____ CCS

Trunk usage

Number of TNsCCS/T perTotal CCS load

Trunk routeaccessing routetrunk routeper trunk route

_____ _____ 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=_____CCS

Digitone receivers
Number of DTRs (from tables)_____
= Total DTR load=_____CCS
=Total load =_____CCS

Worksheet 3: System cabinet/chassis requirements

Customer:_____

Date: _____

Prepare one worksheet for the complete system at cutover, 2-year, and 5-year intervals. Complete Table 94 "IPE card calculations" (page 304) to determine the number of IPE cards required, then go to one of the following, as applicable:

- "Cabinet system calculations without Meridian Mail" (page 305)

- "Cabinet system calculations with Meridian Mail" (page 305)

- "Chassis system calculations" (page 306)

Table 94
IPE card calculations

	Cutover	**2 years**	**5 years**
Number of digital line cards = number of digital ports (M2250 uses 2 ports) 16			
Number of analog line cards = number of analog ports in service 16			
Number of analog waiting line cards = number of analog ports with message waiting 16			
Note: For higher reliability, do not configure more than one M2250 console on one digital line card. Use paging trunks on universal trunk cards or E&M trunk cards, depending on what combination minimizes the total number of trunk cards required.			

	Cutover	**2 years**	**5 years**
Number of universal trunk cards = total number of CO/DID/RAN/paging trunks 8			
Number of E&M trunk cards = total number of E&M/paging/dictation trunks 4			
Number of Voice Gateway Media Cards (Media Card 32-port cards use 1 IPE slot)			
Total cards			

Note: For higher reliability, do not configure more than one M2250 console on one digital line card. Use paging trunks on universal trunk cards or E&M trunk cards, depending on what combination minimizes the total number of trunk cards required.

Cabinet system calculations without Meridian Mail

The first cabinet provides a total of 9 slots for trunk and line cards:

Number of IPE cards	**Number of cabinets required (maximum 5 cabinets)**
1-9	1
10-19	2
20-29	3
30-39	4
40-49	5

For systems requiring SDI/DCH cards, subtract one available card slot from the first cabinet for each additional SDI/DCH card required.

Cabinet system calculations with Meridian Mail

Subtract one available card slot from the first cabinet:

Number of IPE cards	**Number of cabinets required (maximum 5 cabinets)**
1-8	1
9-18	2
19-28	3
29-38	4
39-48	5

Chassis system calculations

The chassis provides a total of 3 locations for trunk and line cards, with the chassis expander providing 4 additional locations:

Number of IPE cards	Number of chassis required (maximum 5 chassis)
1-3	1
4-7	2
Note: If you are adding a Meridian Mail card, it must be located in slot 10 of the chassis expander, which reduces the maximum number of IPE cards to 6.	

For systems requiring extra TDS/DTR or SDI/DCH cards, subtract one available card slot from the chassis for each additional TDS/DTR or SDI/DCH card required.

Number of chassis required:_____

Worksheet 4: Protected memory calculations

Customer: _____

Date: _____

Prepare one worksheet for the complete system.

	Items	Words	Total
Fixed amount of storage required			
500 and 2500 TNs			
Add-on modules			
Trunk units			
Consoles			
Customer groups			
Trunk routes			
Code restricted trunk routes			
DTR loops (in excess on 1)			
Speed call head table			
Speed call lists (10 numbers)			
Speed call lists (50 numbers)			
Note: Record totals on the next page.			

	Items	Words	Total
TDS loops (in excess of 1)			
History file			
Note: Record totals on the next page.			

Total from table   _____

Add 10%   _____

Total words from   _____
table

Capacity   _____ **64** ____ k words (k = 1024 words)

Worksheet 5: Unprotected memory calculations

Customer:_____

Date: _____

Prepare one worksheet for the complete system.

	Items	Words	Total
Fixed amount of storage required			
500 and 2500 TNs			
Add-on modules			
Network groups	2		
Trunk units			
Consoles			
Customer groups			
Network loops	30		
Peripheral Signalling	2		
Trunk routes			
SDI cards			
TDS loops			
Conference loops	3		
DTR loops			
Call registers			

	Items	Words	Total
Low priority input buffers			
High priority input buffers			

Total from table   _____

Total words from   _____
table

Capacity   _____ **64** ___k words (k = 1024 words)

Worksheet 6: Equipment summary

Customer:_____

Date: _____

Prepare one worksheet for the complete system.

Equipment summary	Quantity	Based on
Line and trunk cards		Cutover
DTRs		2 years
TDS loops		2 years
Call registers		2 years
High priority input buffers		Cutover
Low priority input buffers		Cutover
System cabinets		2 years

Worksheet 7: Cabinet system power consumption

Prepare one worksheet for each cabinet in the system. Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption of each circuit card.

Worksheet 7a: Main Cabinet

Slot	Circuit card	Type	Power consumption (W)*
0	NTDK20	SSC	15
1			
2			
3			
4			
5			
6			
7			
8			
9			
10			
		Total	
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Worksheet 7b: First Expansion Cabinet

Slot	Circuit card	Type	Power consumption (W)*
11			
12			
13			
14			
15			
16			
17			
18			
19			
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Slot	Circuit card	Type	Power consumption (W)*
20			
		Total	
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Worksheet 7c: Second Expansion Cabinet

Slot	Circuit card	Type	Power consumption (W)*
21			
22			
23			
24			
25			
26			
27			
28			
29			
30			
		Total	
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Worksheet 7d: Third Expansion Cabinet

Slot	Circuit card	Type	Power consumption (W)*
31			
32			
33			
34			
35			
36			
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Slot	Circuit card	Type	Power consumption (W)*
37			
38			
39			
40			
		Total	

*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.

Worksheet 7e: Fourth Expansion Cabinet

Slot	Circuit card	Type	Power consumption (W)*
41			
42			
43			
44			
45			
46			
47			
48			
49			
50			
		Total	

*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.

Worksheet 7f: Total Cabinet system power consumption

P*out* Main (total for slots 1-10 in Main Cabinet)	
P*out* Expan (total for slots 11-20 in the first Expansion Cabinet)	
P*out* Expan (total for slots 21-30 in the second Expansion Cabinet)	
P*out* Expan (total for slots 31-40 in the third Expansion Cabinet)	
P*out* Expan (total for slots 41-50 in the fourth Expansion Cabinet)	
Total	

Worksheet 8: Chassis system power consumption

Prepare one worksheet for each chassis and chassis expander in the system. Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption of each circuit card.

Worksheet 8a: Main Chassis

Slot	Circuit card	Type	Power consumption (W)*
0	NTDK20	SSC	15
1			
2			
3			
4, 5, 6	NTDK16	48-port DLC	75
		Total	
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Worksheet 8b: Chassis expander

Slot	Circuit card	Type	Power consumption (W)*
7			
8			
9			
10			
		Total	
*Refer to Table 13 "Circuit card and daughterboard power consumption" (page 56) for the power consumption.			

Worksheet 8c: Total Chassis system power consumption

P*out* Main (total for slots 1-6 in the chassis)	
P*out* Expan (total for slots 7-10 in the chassis expander)	
Total	

Note: For an expansion system, use the Cabinet system Worksheets.

Worksheet 9: Battery current and AC line calculation for AC systems using NTAK75 and NTAK76

Main cabinet

PF = 0.6,
V = 110VAC or
208VAC

NTAK75/76
battery unit

VBatt = 48VDC

I *Batt* (Main)

To AC power
source

NTDK70 AC/DC
power supply

Cabinet system
CE & PE

80% efficiency

Note: P*line* (Main) = 750VA +/- 10% maximum during battery charging

P*out* (Main) = _____

P_{in} (Main) = $\dfrac{P_{out}\ (Main)}{0.80}$ = _____

I $Batt$(Main) in DC amps = $\dfrac{P_{in}\ (Main)}{48}$ = _____

I *line* (Main) in AC amps = $\dfrac{P_{in}\ (Main)}{V \times 0.6}$ = _____

Expansion cabinet

PF = 0.6,
V = 110VAC or
208VAC

NTAK75/76
battery unit

VBatt = 48VDC

I *Batt* (Expan)

To AC power
source

NTDK70 AC/DC
power supply

Cabinet system
CE & PE

80% efficiency

Note: P*line* (Expan) = 750VA +/- 10% maximum during battery charging

P*out* (Expan) = _____

P_{in} (Expan) = $\dfrac{P_{out}\ (Expan)}{0.80}$ = _____

I $Batt$ (Expan) in DC amps = $\dfrac{P_{in}\ (Expan)}{48}$ = _____

I *line* (Expan) in AC amps = $\dfrac{P_{in}\ (Expan)}{V \times 0.6}$ = _____

Worksheet 10: Battery current calculation for customer- provided DC reserve power

Main cabinet

V Batt = 48VDC

customer-provided
reserve power
system

I $Batt$ (Main)

NTDK72 DC
power supply

80% efficiency

Cabinet system
CE & PE

P_{out} (Main) = _____

P_{in} (Main) = $\dfrac{P_{out}\text{ (Main)}}{0.80}$ = _____

I $Batt$ (Main) in DC amps = $\dfrac{P_{in}\text{ (Main)}}{48}$ = _____

Expansion cabinet V = 48V

V Batt = 48VDC

customer-provided
reserve power
system

I $Batt$ (Expan)

NTDK72 DC
power supply

80% efficiency

Cabinet system PE

P_{out} (Expan) = _____

P_{in} (Expan) = $\dfrac{P_{out}\text{ (Expan)}}{0.80}$ = _____

I $Batt$ (Expan) in DC amps = $\dfrac{P_{in}\text{ (Expan)}}{48}$ = _____

Figure 43
Discharge time for the NTAK76 Battery

Figure 44
Discharge time for the NTAK75/QBL24A1 Batteries
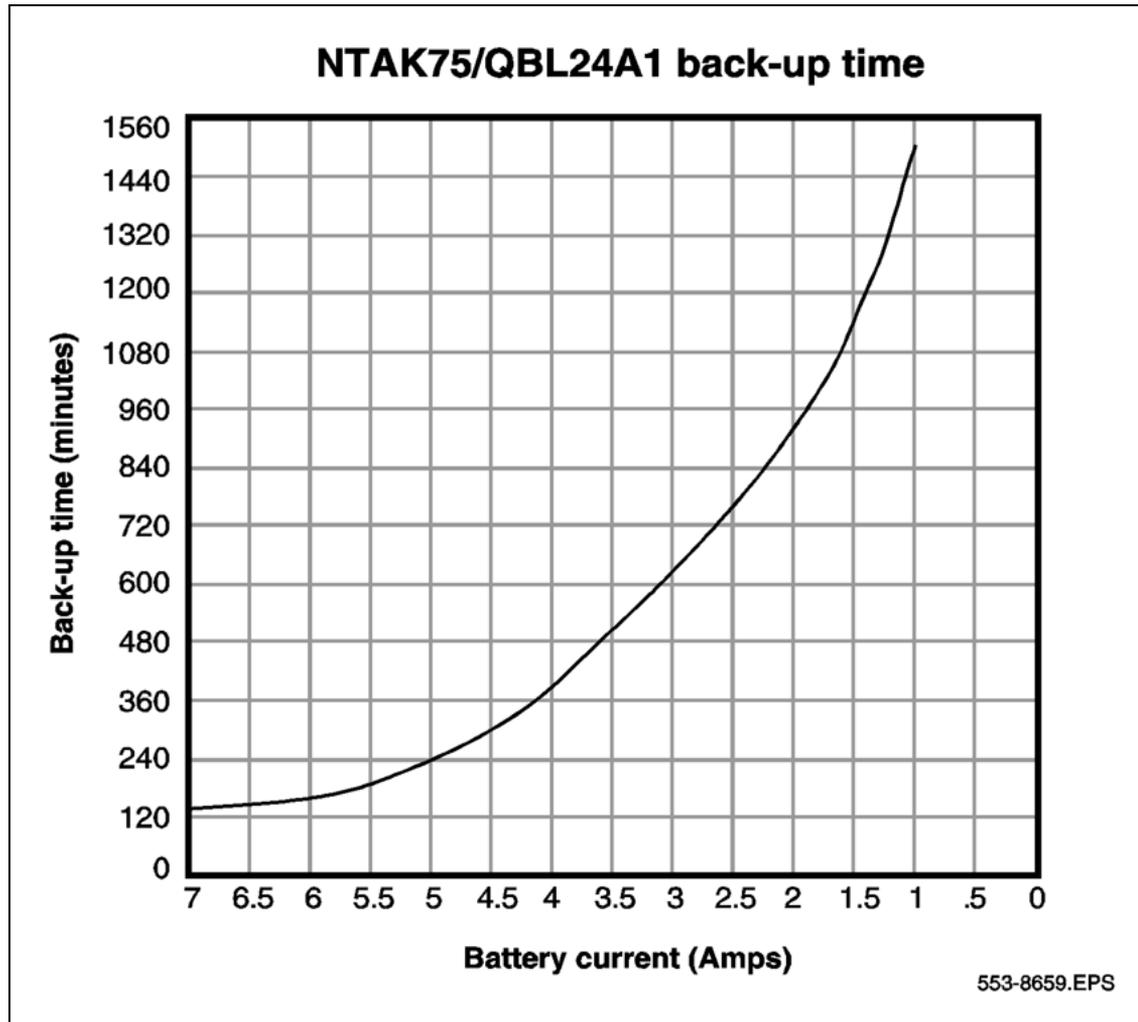
Worksheet 11: Media Card calculation

Input constants	Input configuration data
R_I – intraoffice calls ratio	Number of analog sets
R_T – tandem calls ratio	Number of digital sets
I – incoming calls to total calls ratio	Number of IP Phones
O – outgoing calls to total calls ratio	Number of DECT sets
P – IP calls to total calls ratio	Number of SIP Virtual Trunks (estimated)
V – Virtual Trunk calls to total trunk calls ratio	Number of H.323 Virtual Trunks (estimated)
v_S – SIP Virtual Trunk calls to total Virtual Trunk calls ratio	

Input constants	Input configuration data
v_H – H.323 Virtual Trunk calls to total Virtual Trunk calls ratio	
r_{CON} – Conference loop to traffic loop ratio	
Hold time in seconds (AHT_{XX})for set to set, trunk to trunk, set to trunk, trunk to set	

Worksheet 11a
Media Card calculation procedure

Item	Calculation formula
(1) TDM set CCS	= (Number of analog sets + Number of digital sets) × _____ CCS/set
(2) IP set CCS	= Number of IP sets × _____ CCS/set
(3) DECT set CCS	= Number of DECT sets × _____ CCS/set
(4) Total line CCS	= (1) + (2) + (3)
(5) TDM trunk CCS	= Number of TDM trunks × ____ CCS/trunk
(6) SIP Virtual Trunk CCS	= Number of SIP Virtual Trunks × ____ CCS/trunk
(7) H.323 Virtual Trunk CCS	= Number of H.323 Virtual Trunks × ____ CCS/trunk
(8) Total trunk CCS	= (5) + (6) + (7)
(9) Total system CCS (TCCS)	= (4) + (8)
(10) Weighted average holding time (WAHT)	= $(R_I \times AHT_{SS}) + (R_T \times AHT_{TT}) + (I \times AHT_{TS}) + (O \times AHT_{ST})$
(11) Total calls (T_{CALL})	= $0.5 \times T_{CCS} \times 100 \div WAHT$
(12) Calls requiring DSP resources (C_{DSP})	= (a) + (b) + (c) + (d) + (e) + (f)
— (a) Intraoffice IP-TDM set calls	= $T_{CALL} \times R_I \times 2 \times P \times (1 - P)$
— (b) Tandem VT-TDM trunk calls	= $T_{CALL} \times R_T \times 2 \times V \times (1 - V)$
— (c) IP set to TDM trunk calls	= $T_{CALL} \times O \times P \times (1 - V)$
— (d) TDM set to VT calls	= $T_{CALL} \times O \times (1 - P) \times V$
— (e) VT trunk to TDM set calls	= $T_{CALL} \times I \times V \times (1 - P)$
— (f) TDM trunk to IP set calls	= $T_{CALL} \times I \times (1 - V) \times P$
(13) DSP CCS (CCS_{DSP})	= $C_{DSP} \times WAHT \div 100$
(14) Number of Media Cards required for general traffic	= $CCS_{DSP} \div 794$
(15) DSP channels for Conference	= Total number of sets × P × r_{CON} × 0.4
(16) DSP channels for applications	= (a) + (b) + (c) + (d) + (e) + (f) + (g) + (h)
— (a) CallPilot	= Number of CallPilot ports × P
— (b) MIRAN	= Number of MIRAN ports × P

Item	Calculation formula
— (c) MICB	= Number of MICB ports × P
— (d) MIPCB	= Number of MIPCB ports × P
— (e) MICA	= Number of MICA ports × P
— (f) MIVS	= Number of MIVS ports × P
— (g) BRI	= Number of SILC cards × 16 × P
— (h) Agent greeting ports	= Number of Agent greeting ports × P
(17) Total DSP channels	= [(14) × 32] + (15) + (16)
(18) Total Media Cards	= Roundup((17) ÷ 32)

Worksheet 11b
Virtual Trunk calculation

Call type	Calculation formula
(1) Virtual Trunk calls (C_{VT})	(a) + (b) + (c) + (d) + (e) + (f)
— (a) Tandem VT to TDM trunk calls	$T_{CALL} \times R_T \times 2 \times V \times (1 - V)$
— (b) IP to VT calls	$T_{CALL} \times O \times P \times V$
— (c) TDM set to VT calls	$T_{CALL} \times O \times (1 - P) \times V$
— (d) VT to TDM set calls	$T_{CALL} \times I \times V \times (1 - P)$
— (e) VT to IP set calls	$T_{CALL} \times I \times V \times P$
— (f) Tandem VT (H.323) to VT (SIP) calls	$T_{CALL} \times R_T \times V^2 \times v_H \times v_S \times 2 \times 2$
(2) SIP Virtual Trunk calls	$C_{VT} \times v_S$
(3) H.323 Virtual Trunk calls	$C_{VT} \times v_H$
(4) Virtual Trunk CCS (CCS_{VT})	$C_{VT} \times WAHT \div 100$
(5) Number of Virtual Trunks	Roundup($CCS_{VT} \div 794 \times 32$)
(6) Virtual Trunk traffic in erlangs	Roundup($CCS_{VT} \div 36$) use this for LAN/WAN bandwidth calculation

Note: If the calculated number of Virtual Trunks differs significantly from the original estimated number of Virtual Trunks (> 20%), Nortel recommends using the calculated Virtual Trunk number and repeating the calculation procedure to yield a more accurate number for required Media Cards and Virtual Trunks.

Worksheet 12: Signaling Server calculation

User input (modified if necessary from previous calculations)

Number of IP Phones in the system (IPL) _____

Number of Virtual Trunks _____

User input (modified if necessary from previous calculations)

- Number of SIP Virtual Trunks   _____
- Number of H.323 Virtual Trunks   _____

Number of calls involving at least one IP set (C_{IP})   _____

Number of calls involving Virtual Trunks (C_{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 sets   Yes/No

PD/CL/RL feature sharing database with other traffic   Yes/No

Worksheet 12
Signaling Server calculations

Item	Calculation
(1) Signaling Server for PD/CL/RL database (SSDB)	= a if no PD/CL/RL feature = b if yes on feature, and sharing functions = c if yes on feature, and a stand-alone database
(2) Network Routing Service calculation (SSNR)	= largest of 2(a), (b), or (c) = _____
(a) Endpoints	= NRS endpoints (NRE) ÷ NRS endpoints limit (NRE1) = NRE ÷ 5000 = _____
(b) Dial plan entries	= Dial plan entries (NRD) ÷ NRS endpoints limit (NRD_1) = NRD ÷ 20 000 = _____

Item	Calculation
(c) Calls per hour $NRC_{NET} = VT_{NET} \times$ (CCS per VT) $\times 100 \div$ WAHT $\div 2 =$ _____ SIP calls = (Number of SIP Virtual Trunks) $\times$ (CCS per VT) $\times 100 \div$ WAHT = _____ H.323 calls = (Number of H.323 Virtual Trunks) $\times$ (CCS per VT) $\times 100 \div$ WAHT = _____	= [Local calls (NRC_0) + Network calls (NRC_{NET})] $\div$ NRS calls per hour limit (NRC_{HL}) = [SIP calls + (1.5 $\times$ H.323 calls) + NRC_{NET}] $\div$ 100 000 = _____
Signaling Server requirement for NRS (SSNW) If NRS is in a dedicated Signaling Server, round up SSNR before proceeding.	= ROUNDUP(SSNR) $\times$ NRA (= 2 if true; else = 1) = _____
(3) Terminal Proxy Server calculation (SSTR)	= larger of 3(a), (b), or (c) = _____
(a) If SSDB = a or c	= IPL $\div IPL_{SL}$ = IPL $\div$ 5000 = _____
(b) If SSDB = b • If IPL <= 1000 • If IPL > 1000	 = IPL $\div IPL_{DB}$ = IPL $\div$ 1000 = _____ = 1 + [(IPL $- IPL_{DB}$) $\div IPL_{SL}$] = 1 + [(IPL $-$ 1000) $\div$ 5000] = _____
(c) IP calls	= IPC $\div IPC_{HL}$ = $C_{IP} \div$ 15 000 = _____
Signaling Server requirement for TPS (SSTW) If TPS is in a dedicated Signaling Server, round up SSTR before proceeding.	= ROUNDUP(SSTR) + TPSA (= 1 if true; else = 0) = _____
(4) H.323 Gateway calculation (SSHR)	= larger of 4(a) or (b) = _____

Item	Calculation
(a) H.323 Virtual Trunks	$= HVT \div HVT_{SL} = HVT \div 1200$ = _____
(b) H.323 calls	$= C_{VT} \div HVTC_{HL} = C_{VT} \div 18\ 000$ = _____
Signaling Server requirement for H.323 Gateway (SSHW) If H.323 Gateway is in a dedicated Signaling Server, round up SSHR before proceeding.	$= ROUNDUP(SSHR) \times GWA$ (= 2 if true; else = 1) = _____
(5) SIP Gateway calculation (SSSR)	= larger of 5(a) or (b) = _____
(a) SIP Virtual Trunks	$= SVT \div SVT_{SL} = SVT \div 1800$ = _____
(b) SIP calls	$= C_{VT} \div SVTC_{HL} = C_{VT} \div 27\ 000$ = _____
Signaling Server requirement for SIP Gateway (SSSW) If SIP Gateway is in a dedicated Signaling Server, round up SSSR before proceeding.	$= ROUNDUP(SSSR) \times GSA$ (= 2 if true; else = 1) = _____
(6) Total Signaling Server requirements (SST)	= evaluate in order,
(a) If (SSNR + SSTR + SSHR + SSSR) < 1	= ROUNDUP(SSNR + SSTR + SSHR + SSSR) + (1 if NRA, GWA, GSA, or TPSA true; else 0) + (1 if SSDB = c; else 0)
(b) If (SSTR + SSHR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1	= SSNW + [ROUNDUP(SSTR + SSHR + SSSR) × (2 if GWA, GSA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)
(c) If (SSNR + SSHR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1	= SSTW + [ROUNDUP(SSNR + SSHR + SSSR) × (2 if NRA, GWA, or GSA true; else 1)] + (1 if SSDB = c; else 0)
(d) If (SSTR + SSNR + SSSR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1	= SSHW + [ROUNDUP(SSTR + SSNR + SSSR) × (2 if NRA, GSA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)
(e) If (SSTR + SSNR + SSHR) < 1 and (SSNR + SSTR + SSHR + SSSR) > 1	= SSSW + [ROUNDUP(SSTR + SSNR + SSHR) × (2 if NRA, GWA, or TPSA true; else 1)] + (1 if SSDB = c; else 0)

Item	Calculation
[If the process reaches this step, it means that (SSGR + SSTR + SSHR + SSSR) > 1 and also that there is no combination of three functions that can share one Signaling Server. The algorithm continues by evaluating whether there are combinations of two functions that can share a Signaling Server.]	
(f) If (SSNR + SSTR) < 1 and (SSHR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR + SSTR) × (2 if NRA or TPSA true; else 1)] + [ROUNDUP(SSHR + SSSR) × (2 if GWA or GSA true; else 1)] + (1 if SSDB = c; else 0)
(g) If (SSNR + SSHR) < 1 and (SSTR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR + SSHR) × (2 if NRA or GWA true; else 1)] + [ROUNDUP(SSTR + SSSR) × (2 if TPSA or GSA true; else 1)] + (1 if SSDB = c; else 0)
(h) If (SSNR + SSSR) < 1 and (SSTR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR + SSSR) × (2 if NRA or GSA true; else 1)] + [ROUNDUP(SSTR + SSHR) × (2 if TPSA or GWA true; else 1)] + (1 if SSDB = c; else 0)
(i) If (SSTR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR) × (2 if NRA true; else 1)] + [ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSTR + SSHR) × (2 if TPSA or GWA true; else 1)] + (1 if SSDB = c; else 0)
(j) If (SSNR + SSTR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSHR) × (2 if GWA true; else 1)] + [ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSNR + SSTR) × (2 if NRA or TPSA true; else 1)] + (1 if SSDB = c; else 0)
(k) If (SSNR + SSHR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSTR) + (1 if TPSA true; else 0)] + [ROUNDUP(SSSR) × (2 if GSA true; else 1)] + [ROUNDUP(SSNR + SSHR) × (2 if NRA or GWA true; else 1)] + (1 if SSDB = c; else 0)
(l) If (SSNR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSTR) + (1 if TPSA true; else 0)] + [ROUNDUP(SSHR) × (2 if GWA true; else 1)] + [ROUNDUP(SSNR + SSSR) × (2 if NRA or GSA true; else 1)] + (1 if SSDB = c; else 0)
(m) If (SSTR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR) × (2 if NRA true; else 1)] + [ROUNDUP(SSHR) × (2 if GWA true; else 1)] + [ROUNDUP(SSTR + SSSR) × (2 if TPSA or GSA true; else 1)] + (1 if SSDB = c; else 0)

Item	Calculation
(n) If (SSHR + SSSR) < 1 and (SSGR + SSTR + SSHR + SSSR) > 1	= [ROUNDUP(SSNR) × (2 if NRA true; else 1] + ROUNDUP(SSTR) + (1 if TPSA true; else 0) + [ROUNDUP(SSHR + SSSR) × (2 if GWA or GSA true; else 1)] + (1 if SSDB = c; else 0)
If the scenario has not fallen into any of the above 14 cases, it is not possible to share Signaling Server functions. Use this equation to calculate the total Signaling Server requirement (SST).	= ROUNDUP(SSNW + SSTW + SSHW + SSSW) + (1 if SSDB = c; else 0) where **SSNW** = SSNR(ROUNDUP if dedicated) + [ROUNDUP(SSNR) × (1 if NRA true; else 0)] **SSTW** = SSTR(ROUNDUP if dedicated) + (1 if TPSA true; else 0) **SSHW** = SSHR(ROUNDUP if dedicated) + [ROUNDUP(SSHR) × (1 if GWA true; else 0)] **SSSW** = SSSR(ROUNDUP if dedicated) + [ROUNDUP(SSSR) × (1 if GSA true; else 0)]

Worksheet 13: Real-time calculation

Worksheet 13
Real-time calculation with IP/VT applications

Input constants	Input configuration data
From Worksheet for DSP calculation	
Total calls (TCALL)	= $0.5 \times TCCS \times 100 \div WAHT$
Intraoffice IP-IP calls	= $TCALL \times R_I \times P^2$
Penetration factor pf1	= $R_I \times P^2$
Intraoffice IP-TDM calls	= $TCALL \times R_I \times 2 \times P \times (1 - P)$
Penetration factor pf2	= $R_I \times 2 \times P \times (1 - P)$
Intraoffice TDM-TDM calls	= $TCALL \times R_I \times (1 - P)^2$
Penetration factor pf3	= $R_I \times (1 - P)^2$
Tandem VT - TDM calls	= $TCALL \times R_T \times 2 \times V \times (1 - V)$
Penetration factor pf4	= $R_T \times 2 \times V \times (1 - V)$
Tandem TDM - TDM calls	= $TCALL \times R_T \times (1 - V)^2$
Penetration factor pf5	= $R_T \times (1 - V)^2$
IP set- VT calls	= $TCALL \times O \times P \times V$
Penetration factor pf6	= $O \times P \times V$

Input constants	Input configuration data
IP set- TDM calls	$= \text{TCALL} \times O \times P \times (1 - V)$
Penetration factor pf7	$= O \times P \times (1 - V)$
TDM set- VT calls	$= \text{TCALL} \times O \times (1 - P) \times V$
Penetration factor pf8	$= O \times (1 - P) \times V$
TDM set - TDM trunk calls	$= \text{TCALL} \times O \times (1 - P) \times (1 - V)$
Penetration factor pf9	$= O \times (1 - P) \times (1 - V)$
VT to TDM set calls	$= \text{TCALL} \times I \times V \times (1 - P)$
Penetration factor pf10	$= I \times V \times (1 - P)$
VT -IP set calls	$= \text{TCALL} \times I \times V \times P$
Penetration factor pf11	$= I \times V \times P$
TDM trunk - IP set calls	$= \text{TCALL} \times I \times (1 - V) \times P$
Penetration factor pf12	$= I \times (1 - V) \times P$
TDM trunk - TDM set calls	$= \text{TCALL} \times I \times (1 - V) \times (1 - P)$
Penetration factor pf13	$= I \times (1 - V) \times (1 - P)$
Weighted average penetration factor (PF)	$= (f1 \times pf1) + (f2 \times pf2) + ... + (f12 \times pf12) + (f13 \times pf13)$
Error term (basic features: forward/transfer/conference /waiting)	0.2
Symposium real-time impact (EBCsym)	$= \text{Symposium calls} \times 5.74$
CallPilot real-time impact (EBCcp)	$= \text{CallPilot calls} \times 1.66$
System real-time EBC (System EBC)	$= \text{TCALL} \times (1 + PF + \text{error term}) + EBC_{SYM} + EBC_{CP} + EBC \text{ (other major applications)}$
CPU utilization in % CS 1000M Chassis, CS 1000M Cabinet	$= \text{System EBC} \div 35\,000 \times 100$ for CS 1000M with IP expansion (Chassis/Cabinet) or $= \text{System EBC} \div 42\,000 \times 100$ for CS 1000M without IP expansion (Chassis/Cabinet)

Appendix B
Protected memory requirements

Contents

This section contains information on the following topics:

Introduction

The memory calculations described in "Memory-related parameters" (page 150) and utilized in Worksheet "Worksheet 4: Protected memory calculations" (page 306) are based on assumptions about typical configurations, feature usage, and traffic patterns. This section provides the information required for detailed calculations in cases where those assumptions may not apply.

Protected Memory for Phone Sets: Detail

The protected data blocks for the various set types use varying amounts of memory according to what keys/features are configured on the set. The memory requirements shown in "Memory-related parameters" (page 150) show only a "typical" (as determined by looking at a sampling of sites) size for the given set type. The tables below can be used to arrive at a precise memory requirement if the details of the feature configurations are known. The maximum size permitted for any set's protected data block is 512 words.

PBX sets

The size of the protected line block for PBX sets is determined from the information in Table 95 "Size of protected line block for PBX sets (Units in SL-1 words)" (page 326) (sizes are in SL-1 words).

Table 95
Size of protected line block for PBX sets (Units in SL-1 words)

Feature	SL-1 Compool Variable(s)	Units
Basic Line Block	PPBXBLOCK (words 0-23)	24
Template Area	PBX_TEMPL_AREA (words 24-511 of PPBXBLOCK)	0-487
Card Block Component	1/4 PCARDBLOCK (= 10/4)	2.50

The key layout portion of the template requires:

$(4 + nf) \div rs$ words

where "nf" is the number of features defined for the set, and "rs" is the number of sets sharing the same template.

In addition to the basic line block, each feature requires extra data space as shown in Table 96 "Data space requirements for PBX set features (units in SL-1 words)" (page 326) (sizes are in SL-1 words).

Table 96
Data space requirements for PBX set features (units in SL-1 words)

Feature	SL-1 Compool Variable(s) and/or comment	Units
ACD	P_ACD_KEY_DATA	17
Associate Set (AST)		2
Authcode	.AUTH_TEMPL_SIZE = .NAUT_MAX(6) * (((.AUTH_LEN_MAX(14) - 1)>>2) + 1)	6-24
Automatic Wakeup	HM_STRUCT	8
Call Forward Number	CFW_STRUC (4-24 digits/4)	1-8
Call Park	CALL_PARK_STRUC	2
Call Party Name Display	PBX_NAME_ENTRY	1
CFCT		2
CFNA/Hunting Number	CFNA_ENTRY	4
Dial Intercom Group	PBX_DIG_STRUC	2
DN	PBX_DN_STRUC	3
EFD DN	EFD_STRUC	4
EHT DN	EHT_STRUC	4

Feature	SL-1 Compool Variable(s) and/or comment	Units
Enhanced Hot Line DN	((Number of digits in DN) ÷ 4) + 1 : 4 + 36 digits	2-10
FAXS	FAXS_BLK	17
FFC SCP PASS	FFC_SCPW_STRUC	2
Hot Line DN	((Number of digits in DN) ÷ 4) + 1 : 4 - 36 digits	2-10
HUNT	HUNT_STRUC	4
Internal Call Forward		19
Last Number Redial	Number of digits in LNR DN ÷ 4 : (4 - .MAX_LNR_SIZE = 32) ÷ 4	1-8
Manual Line		2
Message Center DN		2
Message Registration	MR_SET_METER	1
Offhook Interdigit Index	OHAS_INDEXES	1
Pretranslation Enhancement	1/2 word (for 255 calling groups)	1/2
SCI/CCOS/RMS		2
Speed Call Controller	SPEED_CALL_STRUC	1
Speed Call User	SPEED_CALL_STRUC	1
Stored Number Redial	# digits in SNR DN / 4 : (4 − .MAX_SNR_DIGITS = 32) ÷ 4	1-8
System Speed Call User	SPEED_CALL_STRUC	1
Tenant Number	TENANT_NUMBER	1

Digital sets

For digital sets, the requirement is as follows:

- M2006 = 10 + (number of non-key features) ÷ rs
- M2008 = 10 + (number of non-key features) ÷ rs
- M2216 = 20 + 30 × (number of AOM) + (number of non-key features) ÷ rs
- M2616 = 20 + 30 × (number of AOM) + (number of non-key features) ÷ rs
- M2317 = 34 + (number of non-key features) ÷ rs
- M3900 = 34 + (number of non-key features) ÷ rs
- IP Phone = 20 + (number of non-key features) ÷ rs
- IP Softphone 2050 = 20 + (number of non-key features) ÷ rs

where:

- rs is the number of sets sharing the same template

- number of AOM is the number of add-on modules

In addition to the basic line block requirement, each feature requires extra data space as shown in Table 97 "Data space requirements for Meridian 1 set features (units in SL-1 words)" (page 328) (units are expressed in SL-1 words):

Table 97
Data space requirements for Meridian 1 set features (units in SL-1 words)

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
ACD Agent and ID Key	.acd_agent p_acd_key_data KEY xx ACD xxxx(xxx)* yyyy(yyy) *(xxx) - up to 7 digs w/DNXP pkg	17
ACD Display Calls Waiting Key	acd_dwc_ext KEY xx DWC yyyy(yyy)	2
ACD Agent Key (for supervisor)	acd_agt_ext KEY xx AGT yyyy(yyy)	2
ACD Enable Interflow Key	acd_eni_ext KEY xx ENI yyyy(yyy)	2
ACD Night Service DN	acd_nsvc_struc KEY xx NSVC yyyy(yyy)	2
Associate Set (AST)	bcs_ast_struc AST xx yy	3
Authcode (non-key)	.auth_templ_size (6) * (((.AUTH_LEN_MAX (14) - 1)>>2)+1) AUTH n xxxx	6-24
Autodial Key	(4-32 digits/4) .max_adl_size = 31 KEY xx ADL yy (zzzz)	1-8

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
Busy/Forward Status Key	bfs_struct KEY xx BFS tn	1
Call Forward Key	cfw_struc : (.cfw_default (4) or (.MAX_CFW_SIZE=31 + 1)digits/4)	1-8
No Hold Conference and Autodial	(same as autodial) KEY xx CA yy zzzz	1-8
No Hold Conference and Direct Hotline	(htl_dn_size + 3 >>2) + wordoffset(bcs_hot_ter_dn) = (3:34)>>2 + 4 = 4-12 KEY xx CH D yy xxxx	4-12
No Hold Conference and Hotline List	wordoffset(bcs_hot_ter_dn) = 4 KEY xx CH L yyyy	4
No Hold Conference and Speed Call	speed_call_struc KEY xx CS yyyy	1
Dial Intercom Group Key	bcs_dig_struc KEY xx DIG xxxx yy R/V	2
DID Route Control	BCS_DRC_STRUC KEY xx DRC yyy	1
Group Call Key	bcs_grcal_entry KEY xx GRC yy	1
Hotline - One Way, Two Way or Intercom	(htl_dn_size + 3 >> 2) + wordoffset(bcs_hot_ter_dn) = 3:34>>2 + 4 = 4-12 KEY xx HOT D dd yyyy(yyy) KEY xx HOT D dd num DN m KEY xx HOT D nn x...x yyyy(yyy) KEY xx HOT I dd num m	4-12

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
Hotline - One Way or Two Way List	wordoffset(bcs_hot_ter_dn) KEY xx HOT L bbb KEY xx HOT L bbb yyyy(yyy)	4
Internal Call Forward Key	.cfw_default (1) or ((#digs(31) - 1)/4 + 1) : max 8 .max_cfw_size=31 KEY xx ICF 4-(16)-31 xxxx	1-8
Loudspeaker	bcs_dn_struc KEY xx LSPK yyyy	4
Multiple Call Non-ringing DN Key	bcs_dn_entry KEY xx MCN yyyy(yyy)	4
Multiple Call Ringing DN Key	bcs_dn_entry KEY xx MCR yyyy(yyy)	4
Message Registration Key	mr_set_meter KEY xx MRK	1
Message Waiting Key	mwc_entry KEY xx MWK yyyy(yyy)	4
Call Park Key	call_park_struc KEY xx PRK	2
Private Line Non-ringing Key	bcs_dn_entry KEY xx PVN yyyy	4
Private Line Ringing Key	bcs_dn_entry KEY xx PVR yyyy	4

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
Stored Number Redial Key	.max_rdl_size (31): (1+#saved dn digs(3-31))/4 + 1 KEY xx RDL (yy)	2-8
Ringing Number Pickup Key	bcs_rnpg_entry KEY xx RNP	1
Radio Paging	bcs_dn_entry KEY xx RPAG	4
Speed Call Controller Key	speed_call_struc KEY xx SCC yyyy	1
Single Call Non-ringing DN	bcs_dn_entry KEY xx SCN yyyy	4
Single Call Ringing DN	bcs_dn_entry KEY xx SCR yyyy	4
Speed Call User Key	speed_call_struc KEY xx SCU yyyy	1
Signaling Key	bcs_dn_entry KEY xx SIG yyyy(yyy)	4
System Speed Call Controller Key	speed_call_struc KEY xx SSC yyyy	1
System Speed Call User Key	speed_call_struc KEY xx SSU uu	1
Voice Call Key	bcs_dn_entry KEY xx VCC yyyy	4
Non-key Features		

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
Data Equipment Mode (flex voice/data tn)	dtm_struc DEM DTE (DCE)	1
Flexible CFNA DN for External Calls	efd_struc EFD xxxx	4
Hunt DN for External Calls	eht_struc EHT xxxx	4
Flexible Call Forward No Answer	afdn_struc FDN xxxx	4
Offhook Alarm Security DN Index for Forced Out ofSservice	ohas_indexes FSVC (0) - 9	1
Hunt DN (chain) for Internal Calls	hunt_struc HUNT xxxx	4
Alternate Hunt DN (chain) for Internal Calls	ahnt_struc AHNT xxxx	4
Alternate Hunt DN for External Calls	aeht_struc AEHT xxxx	4
Alternate Flexible CFNA DN for External Calls	aefd_struc AEFD xxxx	4
Number of Key Lamp Strips	1 word per KLS in range KLS 1-7	1-7
Last Number Redial Size	.lnr_default(4) or ((xx+1)/4) LNRS xx (4-(16)-32)	1-8
Second DN Sharing Voice Mailbox	bcs_dn_struc SECOND_DN xxxx(xxx)	3

Feature	SL-1 Compool Variable(s), service change format, and/or comment	Units
Station Control Password	ffc_scpw_struc SCPW xxxxx	2
Tenant	tenant_number TEN 1-511	1
Template area users for which commands are implicit or entered outside of LD 11		
Ringing Change Key	supp_features	5
Notification Key Lamp	nkl_data	1
Hospitality Data	hsp_set_data	2
Hotel/Motel Info	hm_struct	8
Campon Priorities	povr_struc	1
Sar Group	save_bcs_sgrp	1
Boss-Secretary Filtering - boss	boss_struc	3
Boss-Secretary Filtering - sec'y	sec_struc	1
Call Party Name Display	PBX_NAME_ENTRY	1
FAXS	FAXS_BLK	17
Xdata Unit Downloadable Parameters	xdata_sc_parms	2

Appendix C
Preprogrammed data

Contents

This section contains information on the following topics:

Components of preprogrammed data

The following items are preprogrammed in the Pre-Configured database on the Software Daughterboard:

For detailed information about preprogrammed data, refer to *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.

Model telephones

A model telephone can be thought of as a default set of features and Class of Service assigned to a telephone.

Telephone models simplify telephone installation. During telephone activation, the telephone prompts you to accept a default model. If a model is chosen, all keys are automatically assigned a feature and no further key programming is required. (The extension number is also predefined using the default numbering plan.)

If you do not want to accept the default model, you can create other models by following the procedures in *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.

> *Note:* Off-Premise Station (OPS) telephones do not have their own telephone models. You can, however, create OPS models by entering DD in response to the CDEN prompt in LD 10.

Trunk route data and model trunks

Preprogrammed trunk routes and trunk models simplify trunk installation procedures. A preprogrammed trunk route supports a certain trunk type, has a default access code, and must be assigned a trunk model. A trunk model supports a certain card type, trunk type, and signaling arrangement.

Trunk models are assigned to default trunk routes using the administration telephone. You can create other models by following the procedures in *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.

Numbering plan

The preprogrammed numbering plan automatically assigns default extension numbers to the following (this list may not be representative of all countries):

- Local extension numbers
- Attendant extension
- Night number
- ACD queues
- Meridian Mail and CallPilot extensions
- Call park extensions

If the default numbering plan does not suit this system's needs, you can change it using the procedures in *Communication Server 1000M and Meridian 1 Small System Installation and Commissioning (NN43011-310)*.

SDI network interface

There are three preprogrammed SDI network interfaces on Small Systems. The NTDK20 SSC card provides TTY ports 0, 1, and 2. All three SDI interfaces can be used as either modem or maintenance ports for TTY terminals.

Tone services

The SSC card provides 30 channels of tone and cadence transmission to the system.

The SSC card also provides tone detection. Units 0–7 can be configured to support DTR/XTD. Units 8–15 can also be configured to support DTR/XTD.

Optionally, units 8–11 can be configured to support other tone detection functions in lieu of DTR/XTD on units 8–15. These other tone functions include one of MFC/MFE/MFK5/MFK6/MFR.

LD 56 contains default tables used for tone and cadence generation.

Table 98
LD 56 tone and cadence data

Preconfigured TDS/DTR data	
TDS loop	Channels 1–30
DTR or XTD	Card 0, units 0–7

Benefits of preprogrammed data

The main benefit of preprogrammed data is that it simplifies installation and activation procedures. compares how a task would be performed using preprogrammed data and how it would be performed without preprogrammed data.

Table 99
Benefits of preprogrammed data

Task	Task performed using preprogrammed data	Task performed without using preprogrammed data
Activating telephones	Plug telephone into socket, lift handset, choose model, choose extension.	Enter LD 10 or 11, enter telephone type, specify TN, assign Class of Service, assign a feature to each key on telephone. • LD 10 has approximately 120 prompts. • LD 11 has approximately 160 prompts.
Activating trunks	Use the administration menu to add a trunk: • Enter a route access code. • Enter a TN. • Enter a trunk model.	Enter LD 16, enter trunk type, access code, signaling arrangements. Enter LD 14, enter TN, route member number, signaling arrangements, Class of Service, and so on. • LD 16 has approximately 200 prompts.

Task	Task performed using preprogrammed data	Task performed without using preprogrammed data
		• LD 14 has approximately 50 prompts.
Establishing a numbering plan	No effort required. Default extension numbers become active when telephones are activated. Default plan is sequential.	A numbering plan must be developed to map TNs to DNs.

Appendix D
Reference tables

List of tables

Trunk traffic Erlang B with P.01 Grade-of-Service

Table 100
Trunk traffic Erlang B (P.01)

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	21	462	41	1076	61	1724	81	2387
2	5.4	22	491	42	1108	62	1757	82	2419
3	16.6	23	521	43	1140	63	1789	83	2455
4	31.3	24	550	44	1171	64	1822	84	2488
5	49.0	25	580	45	1203	65	1854	85	2520
6	68.8	26	611	46	1236	66	1886	86	2552
7	90.0	27	641	47	1268	67	1922	87	2588
8	113	28	671	48	1300	68	1955	88	2621
9	136	29	702	49	1332	69	1987	89	2653
10	161	30	732	50	1364	70	2020	90	2689
11	186	31	763	51	1397	71	2052	91	2722
12	212	32	794	52	1429	72	2088	92	2758
13	238	33	825	53	1462	73	2120	93	2790
14	265	34	856	54	1494	74	2153	94	2822
15	292	35	887	55	1526	75	2185	95	2858
16	319	36	918	56	1559	76	2221	96	2891
17	347	37	950	57	1591	77	2254	97	2923
18	376	38	981	58	1624	78	2286	98	2959
19	404	39	1013	59	1656	79	2318	99	2992
20	433	40	1044	60	1688	80	2354	100	3028
101	3060	121	3740	141	4424	161	5119	181	5810
102	3092	122	3776	142	4460	162	5155	182	5843
103	3128	123	3809	143	4493	163	5188	183	5879
104	3161	124	3845	144	4529	164	5224	184	5915
105	3197	125	3877	145	4561	165	5260	185	5974
106	3229	126	3913	146	4597	166	5292	186	5983
107	3265	127	3946	147	4630	167	5328	187	6019
108	3298	128	3982	148	4666	168	5360	188	6052
109	3330	129	4014	149	4702	169	5396	189	6088

Note: For trunk traffic greater than 6469 CCS, allow 32.35 CCS per trunk.

Trun ks	CCS	Trun ks	CCS	Trun ks	CCS	Trun ks	CCS	Trun ks	CCS
110	3366	130	4050	150	4738	170	5429	190	6124
111	3398	131	4082	151	4770	171	5465	191	6156
112	3434	132	4118	152	4806	172	5501	192	6192
113	3467	133	4151	153	4842	173	5533	193	6228
114	3503	134	4187	154	4874	174	5569	194	6260
115	3535	135	4219	155	4910	175	5602	195	6296
116	3571	136	4255	156	4946	176	5638	196	6332
117	3604	137	4288	157	4979	177	5670	197	6365
118	3640	138	4324	158	5015	178	5706	198	6401
119	3672	139	4356	159	5051	179	5738	199	6433
120	3708	140	4392	160	5083	180	5774	200	6469

Note: For trunk traffic greater than 6469 CCS, allow 32.35 CCS per trunk.

Trunk traffic Poisson 1 percent blocking

Table 101
Trunk traffic Poisson 1 percent blocking

Trun ks	CCS	Trun ks	CCS	Trun ks	CCS	Trun ks	CCS	Trun ks	CCS
1	0.4	21	426	41	993	61	1595	81	2215
2	5.4	22	453	42	1023	62	1626	82	2247
3	15.7	23	480	43	1052	63	1657	83	2278
4	29.6	24	507	44	1082	64	1687	84	2310
5	46.1	25	535	45	1112	65	1718	85	2341
6	64	26	562	46	1142	66	1749	86	2373
7	84	27	590	47	1171	67	1780	87	2404
8	105	28	618	48	1201	68	1811	88	2436
9	126	29	647	49	1231	69	1842	89	2467
10	149	30	675	50	1261	70	1873	90	2499
11	172	31	703	51	1291	71	1904	91	2530
12	195	32	732	52	1322	72	1935	92	2563
13	220	33	760	53	1352	73	1966	93	2594
14	244	34	789	54	1382	74	1997	94	2625

Note: For trunk traffic greater than 6068 CCS, allow 30.34 CCS per trunk.

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
15	269	35	818	55	1412	75	2028	95	2657
16	294	36	847	56	1443	76	2059	96	2689
17	320	37	876	57	1473	77	2091	97	2721
18	346	38	905	58	1504	78	2122	98	2752
19	373	39	935	59	1534	79	2153	99	2784
20	399	40	964	60	1565	80	2184	100	2816
101	2847	121	3488	141	4134	161	4786	181	5442
102	2879	122	3520	142	4167	162	4819	182	5475
103	2910	123	3552	143	4199	163	4851	183	5508
104	2942	124	3594	144	4231	164	4884	184	5541
105	2974	125	3616	145	4264	165	4917	185	5574
106	3006	126	3648	146	4297	166	4549	186	5606
107	3038	127	3681	147	4329	167	4982	187	5639
108	3070	128	3713	148	4362	168	5015	188	5672
109	3102	129	3746	149	4395	169	5048	189	5705
110	3135	130	3778	150	4427	170	5081	190	5738
111	3166	131	3810	151	4460	171	5114	191	5771
112	3198	132	3843	152	4492	172	5146	192	5804
113	3230	133	3875	153	4525	173	5179	193	5837
114	3262	134	3907	154	4557	174	5212	194	5871
115	3294	135	3939	155	4590	175	5245	195	5904
116	3326	136	3972	156	4622	176	5277	196	5937
117	3359	137	4004	157	4655	177	5310	197	5969
118	3391	138	4037	158	4686	178	5343	198	6002
119	3424	139	4070	159	4721	179	5376	199	6035
120	3456	140	4102	160	4754	180	5409	200	6068

Note: For trunk traffic greater than 6068 CCS, allow 30.34 CCS per trunk.

Trunk traffic Poisson 2 percent blocking

Table 102
Trunk traffic Poisson 2 percent blocking

Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
1	0.4	31	744	61	1659	91	2611	121	3581
2	7.9	32	773	62	1690	92	2643	122	3614
3	20.9	33	803	63	1722	93	2674	123	3647
4	36.7	34	832	64	1752	94	2706	124	3679
5	55.8	35	862	65	1784	95	2739	125	3712
6	76.0	36	892	66	1816	96	2771	126	3745
7	96.8	37	922	67	1847	97	2803	127	3777
8	119	38	952	68	1878	98	2838	128	3810
9	142	39	982	69	1910	99	2868	129	3843
10	166	40	1012	70	1941	100	2900	130	3875
11	191	41	1042	71	1973	101	2931	131	3910
12	216	42	1072	72	2004	102	2964	132	3941
13	241	43	1103	73	2036	103	2996	133	3974
14	267	44	1133	74	2067	104	3029	134	4007
15	293	45	1164	75	2099	105	3051	135	4039
16	320	46	1194	76	2130	106	3094	136	4072
17	347	47	1225	77	2162	107	3126	137	4105
18	374	48	1255	78	2194	108	3158	138	4138
19	401	49	1286	79	2226	109	3190	139	4171
20	429	50	1317	80	2258	110	3223	140	4204
21	458	51	1348	81	2290	111	3255	141	4237
22	486	52	1374	82	2322	112	3288	142	4269
23	514	53	1352	83	2354	113	3321	143	4302
24	542	54	1441	84	2386	114	3353	144	4335
25	571	55	1472	85	2418	115	3386	145	4368
26	562	56	1503	86	2450	116	3418	146	4401
27	627	57	1534	87	2482	117	3451	147	4434
28	656	58	1565	88	2514	118	3483	148	4467

Note: For trunk traffic greater than 4533 CCS, allow 30.2 CCS per trunk.

Nortel Communication Server 1000
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NN43011-220   01.01   Standard
Release 5.0   30 May 2007

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Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS	Trunks	CCS
29	685	59	1596	89	2546	119	3516	149	4500
30	715	60	1627	90	2578	120	3548	150	4533

Note: For trunk traffic greater than 4533 CCS, allow 30.2 CCS per trunk.

Digitone receiver requirements Model 1

Table 103
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			

Digitone receiver requirements Model 2

Table 104
Digitone receiver requirements Model 2

Number of DTRs	Max. number of Digitone	DTR load (CCS)	Number of DTRs	Max. number of Digitone	DTR load (CCS)

	lines			lines	
2	2	2	17	843	253
3	21	7	18	920	276
4	52	15	19	996	299
5	90	27	20	1076	323
6	134	40	21	1153	346
7	183	55	22	1233	370
8	235	71	23	1316	395
9	293	88	24	1396	419
10	353	107	25	1480	444
11	416	126	26	1563	469
12	483	145	27	1650	495
13	553	166	28	1733	520
14	623	187	29	1816	545
15	693	208	30	1903	571
16	770	231			

Note: See "Step 5: Calculate Digitone receiver requirements" for Model 2 assumptions.

Digitone receiver requirements Model 3

Table 105
Digitone receiver requirements Model 3

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
2	5	2	17	862	319
3	22	9	18	908	336
4	50	19	19	983	364
5	87	33	20	1062	393
6	132	49	21	1135	420
7	180	68	22	1213	449
8	234	88	23	1294	479
9	291	109	24	1375	509
10	353	131	25	1451	537

Note: See "Step 5: Calculate Digitone receiver requirements" for Model 3 assumptions.

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
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			

Note: See "Step 5: Calculate Digitone receiver requirements" for Model 3 assumptions.

Digitone receiver requirements Model 4

Table 106
Digitone receiver requirements Model 4

Number of DTRs	Max. number of Digitone lines	DTR load (CCS)	Number of DTRs	Max. number of Digitone lines	DTR load (CCS)
2	4	2	17	683	253
3	18	7	18	745	276
4	41	15	19	808	299
5	72	27	20	872	323
6	109	40	21	935	346
7	148	55	22	1000	370
8	193	71	23	1067	395
9	240	88	24	1132	419
10	291	107	25	1200	444
11	340	126	26	1267	469
12	391	145	27	1337	495
13	448	166	28	1405	520
14	505	187	29	1472	545
15	562	208	30	1543	571
16	624	231			

Note: See "Step 5: Calculate Digitone receiver requirements " for Model 4 assumptions.

Digitone receiver load capacity 6 to 15-second holding time

Table 107
Digitone receiver load capacity 6 to 15-second holding time

Number of DTRs	Average holding time in seconds									
	6	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
1	0	0	0	0	0	0	0	0	0	0
2	3	2	2	2	2	2	2	2	2	2
3	11	10	10	9	9	9	9	8	8	8
4	24	23	22	21	20	19	19	19	18	18
5	41	39	37	36	35	34	33	33	32	32
6	61	57	55	53	52	50	49	49	48	47
7	83	78	75	73	71	69	68	67	66	65
8	106	101	97	94	91	89	88	86	85	84
9	131	125	120	116	113	111	109	107	106	104
10	157	150	144	140	136	133	131	129	127	126
11	185	176	170	165	161	157	154	152	150	148
12	212	203	196	190	185	182	178	176	173	171
13	241	231	223	216	211	207	203	200	198	196
14	270	259	250	243	237	233	229	225	223	220
15	300	288	278	271	264	259	255	251	248	245
16	339	317	307	298	292	286	282	278	274	271
17	361	346	335	327	320	313	310	306	302	298
18	391	377	365	356	348	342	336	331	327	324
19	422	409	396	386	378	371	364	359	355	351
20	454	438	425	414	405	398	393	388	383	379
21	487	469	455	444	435	427	420	415	410	406
22	517	501	487	475	466	456	449	443	438	434
23	550	531	516	504	494	487	479	472	467	462
24	583	563	547	535	524	515	509	502	497	491
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

Note: Load capacity is measured in CCS.

Number of DTRs	Average holding time in seconds									
	6	**7**	**8**	**9**	**10**	**11**	**12**	**13**	**14**	**15**
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

Note: Load capacity is measured in CCS.

Digitone receiver load capacity 16 to 25-second holding time

Table 108
Digitone receiver load capacity 16 to 25-second holding time

Number of DTRs	Average holding time in seconds									
	16	**17**	**18**	**19**	**20**	**21**	**22**	**23**	**24**	**25**
1	0	0	0	0	0	0	0	0	0	0
2	2	2	2	2	2	2	2	2	2	2
3	8	8	8	8	8	8	8	8	8	8
4	18	18	18	18	18	17	17	17	17	17
5	31	31	31	30	30	30	30	30	30	29
6	47	46	46	45	45	45	45	44	44	44
7	64	63	63	62	62	62	61	61	61	60
8	83	82	82	81	80	80	79	79	79	78
9	103	102	101	100	100	99	99	98	98	97

Note: Load capacity is measured in CCS.

Number of DTRs	Average holding time in seconds									
	16	17	18	19	20	21	22	23	24	25
10	125	123	122	121	121	120	119	119	118	118
11	147	145	144	143	142	141	140	140	139	138
12	170	168	167	166	165	164	163	162	161	160
13	193	192	190	189	188	186	185	184	184	183
14	218	216	214	213	211	210	209	208	207	206
15	243	241	239	237	236	234	233	232	231	230
16	268	266	264	262	260	259	257	256	255	254
17	294	292	290	288	286	284	283	281	280	279
18	322	319	317	314	312	311	309	308	306	305
19	347	344	342	339	337	335	334	332	331	329
20	374	371	368	366	364	361	360	358	356	355
21	402	399	396	393	391	388	386	385	383	381
22	431	427	424	421	419	416	414	412	410	409
23	458	454	451	448	445	442	440	438	436	434
24	486	482	478	475	472	470	467	465	463	461
25	514	510	506	503	500	497	495	492	490	488
26	544	539	535	532	529	526	523	521	518	516
27	573	569	565	561	558	555	552	549	547	545
28	603	598	594	590	587	584	581	578	576	573
29	631	626	622	618	614	611	608	605	602	600
30	660	655	651	646	643	639	636	633	631	628
31	690	685	680	676	672	668	665	662	659	656
32	720	715	710	705	701	698	694	691	688	686
33	751	745	740	735	731	727	724	721	718	715
34	782	776	771	766	761	757	754	750	747	744
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

Note: Load capacity is measured in CCS.

Digitone receiver requirements Poisson 0.1 percent blocking

Table 109
Digitone receiver requirements Poisson 0.1 percent blocking

Number of DTRs	DTR load (CCS)	Number of DTRs	DTR load (CCS)
1	0	26	469
2	2	27	495
3	7	28	520
4	15	29	545
5	27	30	571
6	40	31	597
7	55	32	624
8	71	33	650
9	88	34	676
10	107	35	703
11	126	36	729
12	145	37	756
13	166	38	783
14	187	39	810
15	208	40	837
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 110
Conference and TDS loop requirements

Network loops required at 2 years	TDS loops required	Conference loops required
1–12	1	1
13–24	2	2
25–36	3	3
37–48	4	4
49–60	5	5
61–72	6	6
73–84	7	7
85–96	8	8
97–108	9	9
109–120	10	10

Digitone receiver provisioning

Table 111
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

Note: Provisioning assumes an 11-second holding time.

DTR CCS	DTR ports	DTR CCS	DTR ports
260–286	16	920–951	38
287–313	17	952–984	39
314–342	18	985–1017	40
343–371	19	1018–1050	41
372–398	20	1051–1084	42
399–427	21	1085–1118	43
428–456	22	1119–1153	44
457–487	23	1154–1188	45
1189–1223	46	1961–1995	68
1224–1258	47	1996–2030	69
1259–1293	48	2031–2065	70
1294–1329	49	2066–2100	71
1330–1365	50	2101–2135	72
1366–1400	51	2136–2170	73
1401–1435	52	2171–2205	74
1436–1470	53	2206–2240	75
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

Note: Provisioning assumes an 11-second holding time.

DTR CCS	DTR ports	DTR CCS	DTR ports
2836–2870	93	3046–3080	99
2871–2905	94	3081–3115	100
2906–2940	95	3116–3465	101
Note: Provisioning assumes an 11-second holding time.			

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