Succession 1000

Succession 3.0 Software

Succession 1000 System

Planning and Engineering

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Revision history

October 2003

Standard 1.00. This document is a new NTP for Succession 3.0. It was created to support a restructuring of the Documentation Library. This document contains information previously contained in the following legacy document, now retired: Succession CSE 1000 Planning and Engineering Guidelines (553-3023-102).

Contents

About this document

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

Subject

This document provides the information necessary to properly engineer a Succession 1000 system.

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

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 Meridian Mail and Meridian Link Module, is not the topic of this document. Guidelines for feature and auxiliary platform engineering are given in documents relating to the specific applications involved. Sufficient information is given in this document to determine and account for the impact of such features and applications upon the capacities of the Succession 1000 system itself.

There are two major purposes for using this document: to engineer an entirely new system, and to evaluate a system upgrade. The procedures for these activities are described in this document.

The Meridian Configurator System provides an alternative to the manual process given in this document. It is beyond the scope of this document to describe the Meridian Configurator process.

Note on legacy products and releases

This NTP contains information about systems, components, and features that are compatible with Succession 3.0 Software. For more information on legacy products and releases, click the **Technical Documentation** link under **Support** on the Nortel Networks home page:

http://www.nortelnetworks.com/

Applicable systems

This document applies to the Succession 1000 system.

Intended audience

The primary audience for this document are the system engineers responsible for engineering the switch, and the Nortel Networks Technical Assistance Support personnel who support them. The engineer can be an employee of the end user, a third-party consultant, or a distributor.

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

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

Conventions

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

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Feature Listing* (553-3001-011)
- *Data Networking for Voice over IP* (553-3001-160)
- *Electronic Switched Network: Signaling and Transmission Guidelines* (553-3001-180)
- *Transmission Parameters* (553-3001-182)
- *Dialing Plans: Description* (553-3001-183)
- *Circuit Card: Description and Installation* (553-3001-211)
- *Branch Office* (553-3001-214)
- *System Management* (553-3001-300)
- *Software Input/Output: Administration* (553-3001-311)
- *Optivity Telephony Manager: System Administration* (553-3001-330)
- *Using Optivity Telephony Manager Release 2.1 Telemanagement Applications* (553-3001-331)
- *IP Line: Description, Installation, and Operation* (553-3001-365)
- *Telephones and Consoles: Description* (553-3001-367)
- *Internet Terminals: Description* (553-3001-368)
- *ISDN Primary Rate Interface: Features* (553-3001-369)
- *Basic Network Features* (553-3001-379)
- *ISDN Basic Rate Interface: Features* (553-3001-380)
- *Software Input/Output: Maintenance* (553-3001-511)
- *Large System: Planning and Engineering* (553-3021-120)
- *Succession 1000 System: Overview* (553-3031-010)

Online

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

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CD-ROM

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

Overview

Contents

This section contains information on the following topics:

Introduction

This chapter outlines the topics involved in planning for a Succession 1000 system installation.

Telephony planning

Telephony planning topics in this document are as follows.

Table 1
Telephony planning

Topic	Description
Installation	Lists general procedures and requirements an installation.
Milestones	Charts milestone activities for an installation.
Deployment	Introduces high-level planning concerns: • Evaluating the existing telephony infrastructure. • Evaluating and documenting the existing data infrastructure. • Planning for deployment of the Succession 1000 system.
Numbering plans and call routing	Highlights how various plans relate to the Gatekeeper function. • Numbering plan options. • Call Routing operation. • PBX numbering plan and routing. • Gatekeeper routing.
Zoning plan	Highlights how zones relate to the Gatekeeper operation
DTI/PRI clocking issues	Provides details about clocking related to this system • Synchronization methods. • North American hierarchical synchronization. • Guidelines. • Clock controller function and description. • Succession 1000 clocking operation. • Installation and configuration.

Data network planning for VoIP

Data network planning for Voice over IP (VoIP) topics in this document are as follows.

Table 2
Data network planning

Topic	Description
System network requirements	Outlines ELAN and TLAN essentials.
Basic data network requirements for Succession Call Server to Succession Media Gateway connections	Outlines voice quality concerns about jitter, bandwidth, and LAN recommendations.
Basic data network requirements for Internet Telephones	Outlines bandwidth requirements and planning.
Power requirements for i2004 and i2002 Internet Telephones	Lists power supply transformers required.

Site planning

Site requirement topics in this document are as follows.

Table 3
Site planning (Part 1 of 2)

Topic	Description
Fire protection and safety requirements	Defines fire protection, fire prevention, and security precautions.
Environmental requirements	Outlines general temperature and humidity conditions.
Other environmental factors	Highlights such factors as static, vibration, dust, lighting, and EFI/RFI.
Selecting a site	Delineates space, location, grounding power, and structural integrity.

Table 3
Site planning (Part 2 of 2)

Topic	Description
Developing the site	Defines the equipment room and floor plan.
System VoIP networking	Refers to detailed NTPs. Basic data network requirements for Internet telephones.
Building cable plan for circuit switched equipment	Describes wire routing and termination points.
Preparing for delivery	Explains what to do for delivery.
Preparing for installation	Ensures that all installation points are done.
Grounding requirements	Provides detail about grounding, single point grounding, safety, and related topics, including grounding method and conduit requirements.
Commercial power requirements	Describes for the following: • The ac power installation for systems installed in a rack • Alternative ac-powered installation • Power consumption worksheets for the Succession 1000
Auxiliary equipment power	Specifies wiring conditions for modems, printers, terminals, and data units.
Modem requirements	Refer to the appropriate NTP for detail.
Maintenance and administration terminals	Refer to the appropriate NTP for terminal details. (For example, Remote access, Administration Tools.)
Cross-connect terminal requirements	Refer to the appropriate NTP for detail.

Preparing for regulatory requirements

This document discusses the following topics related to regulatory requirements.

Table 4
Preparing for regulatory requirements

Topic	Description
System approval	Verifies that the system is approved.
Notice for United States installations	Briefs the FCC regulation for the following: • Importance of Ringer Equivalence Number • Hearing aid compatibility
Notice for Canadian installations	Briefs the Industry Canada regulations.
European compliance information	Lists CTR and EN regulation numbers.
Canadian and United States Network connections	Defines FCC compliance: registered equipment for Direct Inward Dial (DID) calls.
Radio and TV interference	Defines information for the United States and Canada.

Reliability strategies

The following are the planning reliability strategies topics outlined in this document.

Table 5
Planning reliability strategies

Topic	Description
Primary Succession Call Server failure	Defines Succession Call Server redundancy survivability.
Network failure	Defines Survivable Succession Media Gateways.
Succession Signaling Server failure	Defines Succession Signaling Server redundancy.
Gatekeeper failure	Defines Gatekeeper redundancy.
Gatekeeper failure - fail-safe	Defines Gatekeeper fail-safe.
Branch Office serviceability	Defines Survival mode operation (Local Mode).
Campus-distributed Succession Media Gateway in Survival Mode	Defines Survivable Succession Media Gateways.

Deployment planning

To properly deploy a Succession 1000 system, you must first evaluate the existing telecom and data infrastructure.

The telephony features offered in the Succession 1000 system are based on the Meridian 1 capabilities. However, Succession 1000 system infrastructure differs from Meridian 1 infrastructure. Feature performance is only as good as the connections from the data network to the system. A technical understanding of data networking and VoIP is essential for optimal performance of the Succession 1000 system.

Telephony planning

Contents

This section contains information on the following topics:

Installation planning

Use Table 6 as a guide to prepare a detailed plan for every installation.

Table 6
installation planning

Procedure	Requirements
Research	Determine requirements for fire protection and safety, the equipment room, grounding and power, and cables.
Site planning	Select a site with suitable qualifications. Develop the site to meet requirements. Prepare the building cabling plan.
Delivery and installation preparation	Perform preinstallation inspections. Examine the delivery route. Review equipment handling precautions. Gather all delivery items.

Milestone chart

Site preparation activities are easier to plan and monitor when a milestone chart is used. A milestone chart is a general schedule that shows all required activities in order, with a start and end date for each. Individual operations and an overall installation schedule should both be represented. Table 7 on page 23 lists typical activities in a milestone chart. For a complex site, a more detailed chart could be required.

Table 7
Milestone chart

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

Evaluating existing telephony infrastructure

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

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

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

Telephony planning

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

- **Desktop features**. For details about desktop features, see the following:
  — *Telephones and Consoles: Description* (553-3001-367)
  — *Internet Terminals: Description* (553-3001-368)
- **System features**. For details about feature operation, see the *Feature Listing* (553-3001-011). For details about feature configuration, see the *Software Input/Output: Administration* (553-3001-311).

- **System interworking and networking**. For details about Numbering/ Dial Plan Configuration, see the following:
  - *Electronic Switched Network: Signaling and Transmission Guidelines* (553-3001-180)
  - *Dialing Plans: Description* (553-3001-183)
  - *Basic Network Features* (553-3001-379)
- **PRI/DTI clocking**. For details about PRI/DTI clocking, see the following:
  - *ISDN Primary Rate Interface: Features* (553-3001-369)
  - *ISDN Basic Rate Interface: Features* (553-3001-380)

Applications

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

- *Automatic Call Distribution: Description* (553-3001-351)
- CallPilot 555-7101- xxx series NTPs
- Symposium 297-2183-xxx series NTPs
- Remote Office 555-8421-xxx series NTPs
- MDECT 553-3601-xxx series NTPs
- other applications NTPs

Access

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

- *ISDN Primary Rate Interface: Features* (553-3001-369)
- *ISDN Basic Rate Interface: Features* (553-3001-380)

For details about FXS, FXO, or ground/loop start COT trunks, see the *Circuit Card: Description and Installation* (553-3001-211).

Numbering plans

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

- "Uniform Dialing Plan (UDP)" on page 26
- "Coordinated Dialing Plan (CDP)" on page 27
- "Transferable Directory Numbers (TNDN)" on page 27

Uniform Dialing Plan (UDP)

According to this numbering plan, each location within the network is assigned a Location Code (LOC). Each telephone has a Directory Number (DN) that is unique within their Succession Call Server (and end-user). To reach a user you must know their Location Code and their DN.

The dialing sequence for a user is:

Network Access Code (AC1) + Location Code + DN

For example, assuming an network access code of AC1 = 6, a location code of 343 and a directory number of 2222 would be dialed as: 6 343 2222.

With a UDP numbering plan, the Gatekeeper must keep the Home Location Code (HLOC) of every Gateway that is registered for UDP routing. To route a call, the H.323 Gateway passes the Location Code and DN to the Gatekeeper to determine the IP addressing information of the desired Gateway. The Gatekeeper searches for the Location Code within its database and returns the IP addressing information for the site, after which the Gateway software can directly set-up a call to the desired Gateway.

Gatekeeper role

The basic role of a Gatekeeper is to perform address translation from an alias (in this case, a telephone number) to a IP signaling address, and to authorize the call in the H.323 network.

Coordinated Dialing Plan (CDP)

According to this numbering plan, each location is allocated one or more "Steering Codes" that are unique within a CDP domain. This enables users to reach Directory Numbers on multiple Succession Call Servers with a short dialing sequence.

> ***Note:*** A steering code is a unique 1-7 digit prefix, which identifies the network node on which an extension is located, followed by the remaining digits that uniquely identify the extension. A steering code cannot be the same as any access code or other extension number.

The DN of each user (including the steering code) must be unique within the CDP domain. For example, a number of Succession Call Servers may be coordinated so five-digit dialing can be performed within a campus environment. The steering code allocation could be as follows:

- Succession Call Server A: Steering codes 3 and 4
  (that is, DN's in the range 3xxxx and 4xxxx)
- Succession Call Server B: Steering code 5
  (that is, DN's in the range 5xxxx)

Within this group of Succession Call Servers, users can reach each other by dialing their unique DNs. However, all DNs on Succession Call Server A must be in the range 3xxxx or 4xxxx, whereas all DNs on Succession Call Server B must be in the range 5xxxx.

> ***Note:*** If a user moves from one Succession Call Server to another, their DN must change in a CDP numbering plan (see "Transferable Directory Numbers (TNDN)" on page 27).

CDP can be used in conjunction with UDP. To reach UDP numbers, dial AC1. To reach CDP DNs, dial within a CDP domain.

Transferable Directory Numbers (TNDN)

According to this numbering plan, each user is given a unique DN, which does not change if they move to a different Succession Call Server. The Gatekeeper must track each TNDN in the network so it knows which Gateway(s) to return when asked to resolve a TNDN address.

Networks that use a TNDN numbering plan enable users to move from location to location while retaining their Directory Number. This capability is provided through the Network Management and call routing capabilities of Succession Call Server software.

Off-Net Call Routing operation

When dialing calls to PSTN interfaces, the Succession Call Server determines that the call is destined to be Off-Net, based upon digit analysis that must be configured at major network Succession Call Servers. This enables Gateway software to request the location of public E.164 numbers from the Gatekeeper. The Gatekeeper is configured with a list of alternate routes that can reach a particular numbers, each of which is configured with a “Gatekeeper Cost Factor” to determine the least-cost route for the call.

When a Gatekeeper replies to a Gateway with the address information for E.164 numbers, it provides a list of alternative gateways, sorted in order of cost. If a Gateway is busy when a call attempt is made, the originating Gateway tries the next alternative in the list. If no alternative is available over the IP network, the originating Succession Call Server can step to the next member of its route list, which could be a PSTN or TIE route. For example, if an IP network outage occurs that does not allow voice calls to terminate over the IP network, calls reroute to alternate PSTN or TIE routes.

> ***Note:*** E.164/International numbers consist of a one-to-three digit Country Code (CC) followed by National (Significant) Number (N[S]N) with a maximum length 15-n digits, where n is the length of the CC.

Call Routing to/from a local Branch Office Gateway

Since Internet Telephone users can reside at a Branch Office with a Branch Office Gateway, call routing to the local gateway is important, especially when toll charges apply to calls made from the central Succession Call Server that controls the telephone. The administrator can configure digit manipulation for Internet Telephones that are located near a Branch Office Gateway. Digit manipulation enables the administrator to select a gateway that provides PSTN access that is local to the telephone.

Calls from the PSTN to users within the network can be routed according to various ESN numbering plan configurations or according to a new feature called Network Number Resolution, which is based upon the Vacant Number Routing feature. Network Number Resolution enables small sites with minimal configurations to route calls through other Succession Call Servers or the Gatekeeper, such as sites that use the Branch Office Gateway. For further details about the Branch Office Gateway, see the *Branch Office* (553-3001-214).

Call routing mechanisms

There are two primary mechanisms used for call routing:

- the PBX numbering plan
- the Gatekeeper

PBX numbering plan and routing

When a user dials a number, the system determines whether the number is internal or external. If the number is internal, the system terminates the call on the appropriate terminal. If the number is external, the system can process the call in one of two ways:

- Use UDP or CDP to route the call to the proper trunk group.
- Use Vacant Number Routing (VNR) to route the call to a Gatekeeper.

With VNR, any number that is not in the system numbering plan routes to a specific trunk route and a Gatekeeper determines the call destination according to information in a centralized database.

Gatekeeper routing

Once the system determines that a call must be sent over an IP network, the call routes to H.323 gateway software, which uses the Gatekeeper to route the call. The basic role of a Gatekeeper is to perform address translation from a telephone number to an IP signaling address, and to authorize the call in a H.323 network.

Zoning plan

In an H.323 network, each Gatekeeper controls one zone. Each H.323 zone can have multiple H.323 IP clients (Internet Telephones) and multiple Voice Gateway Media Cards (VGMC).

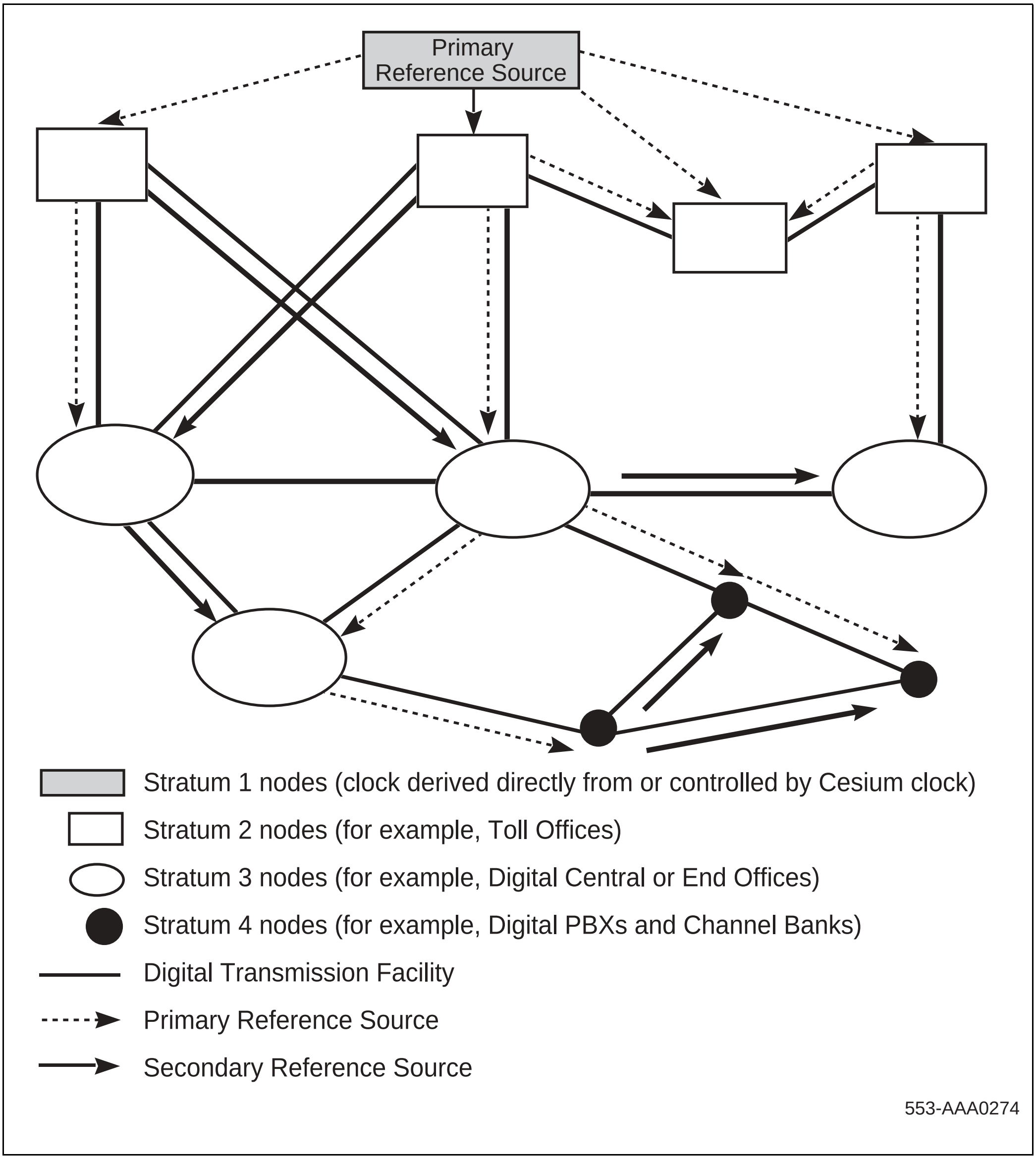
If a call terminates beyond the callers own zone, the gatekeeper in another zone becomes involved in connection set-up. A Succession 1000 network can be divided into several zones. Similarly, an end-user within a system can be partitioned into different zones. However, in most cases, one zone is assigned to one system and one end-user.

DTI/PRI clocking

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

When only two PBX systems interconnect in an isolated private network, the two systems can operate in master-slave mode to achieve synchronization. In master-slave mode, one system derives its timing from the other. Slips can be lessened by forcing all systems to use a common reference clock through a network clocking hierarchy, shown in Figure 1 on page 31.

Figure 1
Hierarchical synchronization



Synchronization methods

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

- Pleisiochoronous operation
- Mesochronous operation

Pleisiochoronous operation

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

Mesochronous operation

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

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

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

North American hierarchical synchronization

Figure 1 on page 31 provides a general view of clock synchronization in a digital network, including the four Stratum levels, where Stratum 1 offers the highest accuracy and Stratum 4, the lowest. Also shown in Figure 1 on page 31 are ways to provide a secondary clock source to prevent timing loops that can cause instability in digital network synchronization.

Timing reference

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

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

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

Node categories/Stratum

In the North America digital network, nodes are synchronized using a priority master/slave method. Digital networks are ranked in Node Categories A to E in Canada, as shown in Table 9 on page 35, and in Stratum levels 1 to 5 in USA. Each node is synchronized to the highest ranking clock where the node has a direct link.

Table 8
Stratum data

	Stratum 2	Stratum 3	Stratum 4
Accuracy	+/- 1.6 * 10^-8 Hz	+/- 4.6 * 10^-6 Hz	+/- 3.2 * 10-5 Hz
Holdover	1 * 10^-10 per day	<=255 frame slips in 1st 24 hours	Not Required
Hardware Duplication	Required	Required (see note 1)	Not Required
MTIE During Rearrangement	MTIE <= 1 usec Phase Change Slope: <= 81 ns in any 1.326 msec	MTIE <= 1 usec Phase Change Slope: <= 81 ns in any 1.326 msec	Not Required (see Note 2)
Pull-in Range	3.2 * 10^-8 Hz	9.2 * 10^-8 Hz	6.4 * 10^-5 Hz
Dedicated Timing Required	Required	Required	Not Required

Note 1: Non-duplicated hardware that meets all other Stratum 3 requirements is referred to as Stratum 3ND.

Note 2: Stratum 4 clock hardware that meets MTIE requirements during rearrangements is referred to as 4E.

Table 9
Node categories

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

Frame slip

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

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

Guidelines

Design guidelines for Succession 1000 Network Synchronization are as follows:

- Where possible, the master Clock Source should always be from a Node Category/Stratum with a higher clock accuracy. When the PBX is connected to the CO, the CO is always the master and the PBX is the slave. For example, the PBX clock controller prompt PREF is set to the slot number of the DTI/PRI connected to the CO.
- Clock controllers within the system should not be in free-run unless they operate in a fully independent network where the source clock controller acts as a master. Only one clock controller in the system can operate in free-run mode.
- When connecting two PBXs together with no CO connections, the most reliable PBX should be the master clock source.
- Avoid timing loops. A timing loop occurs when a clock uses as its reference frequency, a signal that is traceable to the output of the same clock. This produces a closed loop that leads to frequency instability.
- All Central Offices/PBX links that serve as clock references must offer a traceable path back to the same Stratum 1 clock source.
- If a Succession Media Gateway chassis has at least one DTI, PRI, or BRI trunk card, it must also have one clock controller installed. The clock controller tracks to the same traceable reference as the other Succession Media Gateways.
- All slave clock controllers must set their primary reference (PREF) to the slot in which they are installed. For example, a clock controller installed in slot 11 must have its PREF set to slot 11.
- Locate the secondary clock source (SREF) within the same chassis as the primary clock source (PREF).

- The master reference clocking cannot travel through an IP connection within the Succession 1000 system and supply a master reference clock to systems connected with DTI/PRI links. This situation does not provide stable and accurate clocking to remote systems due to jitter and frequency drifting within the Succession Media Gateway.
- The Succession Media Gateway Expansion chassis does not support clock controllers.

Clock controller function and description

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

Clocking modes

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

Tracking mode

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

There are two stages to clock controller tracking:

1 tracking a reference

2 locked onto a reference

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

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

Monitoring references

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

Reference switchover

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

Auto-recovery and chatter

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

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

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

Holdover and free-run

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

Free-run (non-tracking)

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

Faceplate LEDs

Table 10 provides a description of the NTAK20 LEDs.

Table 10
NTAK20 LED indications

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

Clocking operation

The Succession 1000 system can support up to four active clock controllers. However, a Succession Media Gateway chassis can support only one clock controller, and a Succession Media Gateway Expansion chassis cannot support a clock controller. At any moment, the system tracks and locks to any one of the four installed clock controllers within the Succession Media Gateways.

Once the system powers-up and enables all gateways, it searches in a specific order for an active and locked clock controller. The Succession Call Server searches for clock tracking in the Succession Media Gateway chassis according to the order 2, 1, 3, 4. That is, the Succession Call Server locks to the first Succession Media Gateway chassis (Succession Media Gateway chassis 2) in the search order list with a Clock Controller in locked status. If a fault occurs in the link and the clock controller loses lock, the system attempts to synchronize to the chassis' secondary reference clock (SREF) if a secondary is configured. If the secondary reference is not defined, the system attempts to lock on to the next configured clock controller in the search list.

Free-running clocks

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

Figure 2 to Figure 4 show acceptable and unacceptable connections.

Figure 2
Acceptable connection to an isolated private network with primary reference

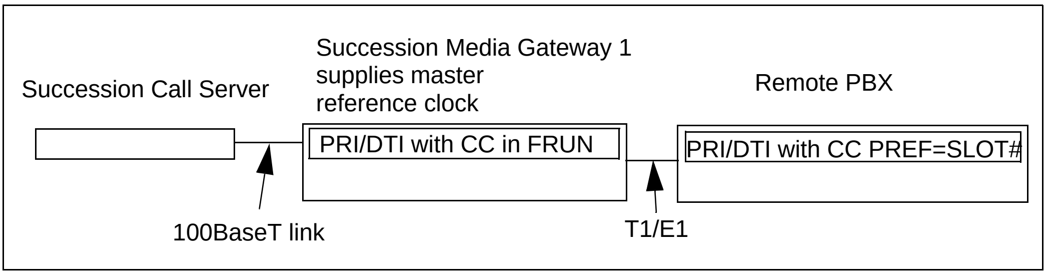


Figure 3
Acceptable connection to an isolated private system with primary and secondary reference

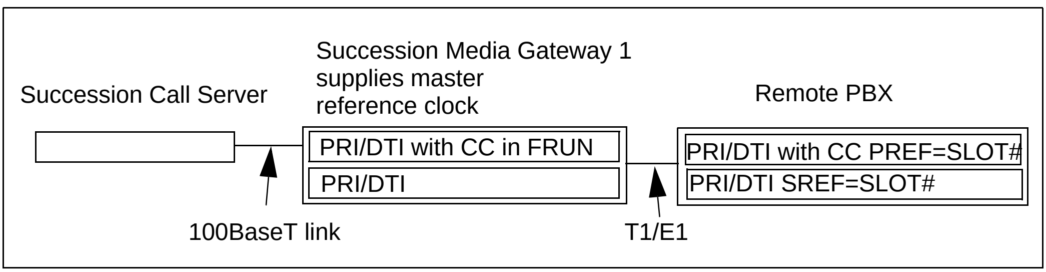
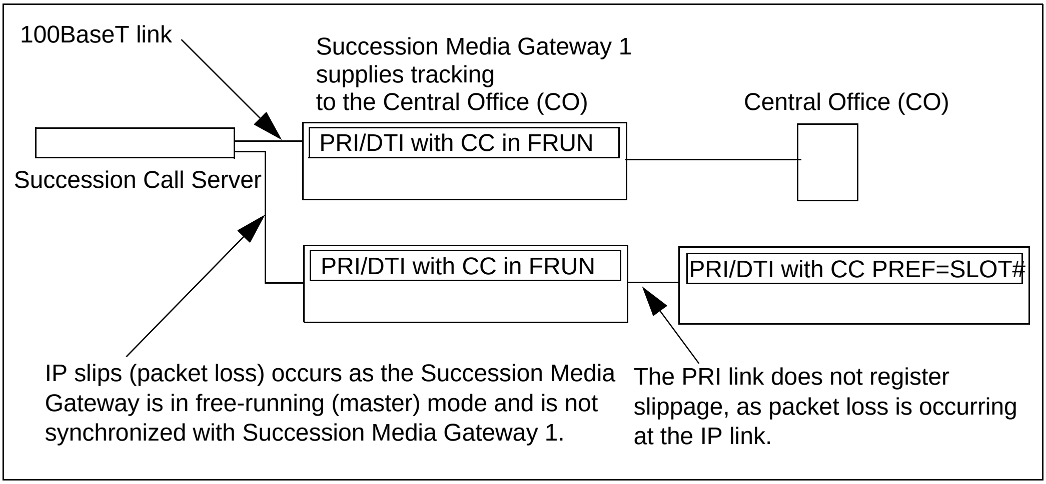


Figure 4
Unacceptable connection:
Succession Media Gateway 1 and Succession Media Gateway 2 do not synchronize



Connecting to a CO

Any Succession Media Gateway chassis that supplies a reference to a remote PBX must have a trunk tracking to a CO, unless the system is in a private network. It is preferable for a digital trunk connected to a remote system to not reference a CO. The digital trunk should be installed in the same cabinet with the CO connection (see Figure 5 on page 43). This configuration is better in situations where the system goes into survivability mode, and connection to the CO is maintained from the remote PBX.

> ***Note:*** If a Succession Media Gateway chassis receives clocking (PREF) from a CO, you cannot have a free-running clock controller anywhere within the Succession 1000 system.

Figure 5
Acceptable connection:
Succession Media Gateway 1 and Succession Media Gateway 2 receive clock reference directly from CO

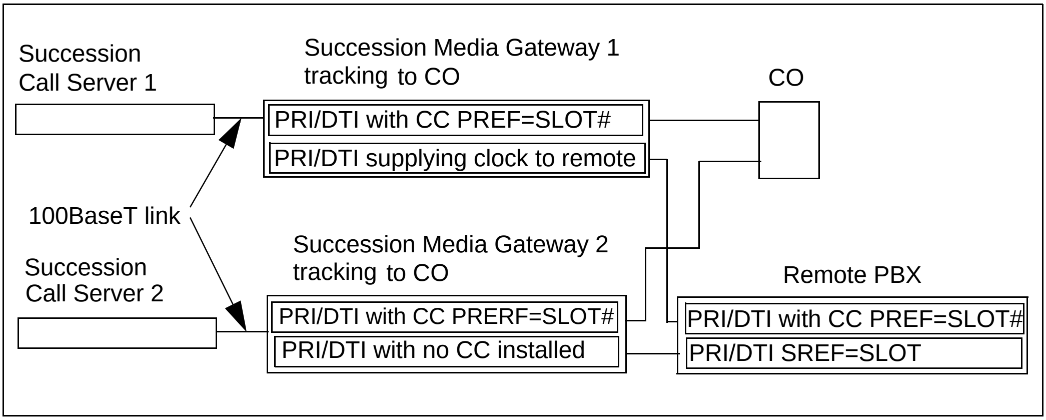


Figure 6
Acceptable connection:
Succession Media Gateway 1 receives clock reference directly from CO / Remote and Succession Media Gateway 2 receives clock reference indirectly from CO

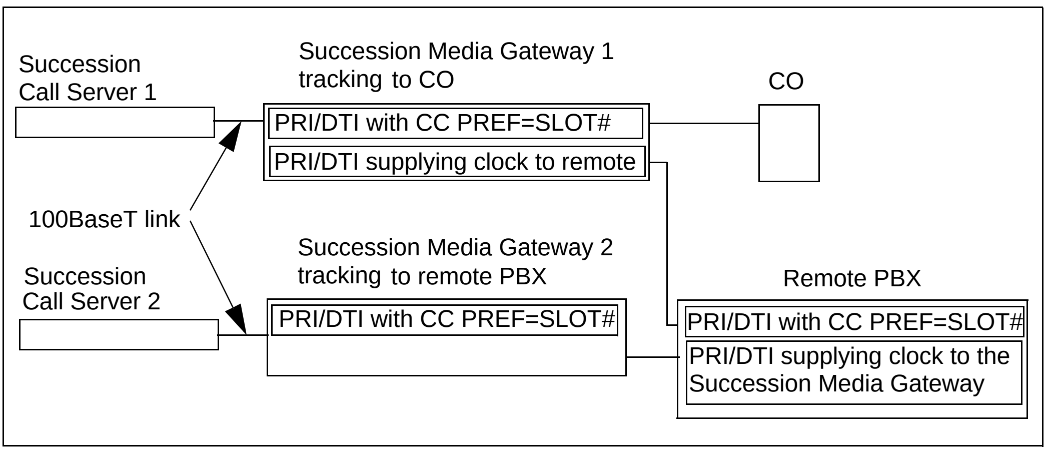


Figure 7
Acceptable connection:
Succession Media Gateway 2 receives clock reference from remote PBX referenced to CO

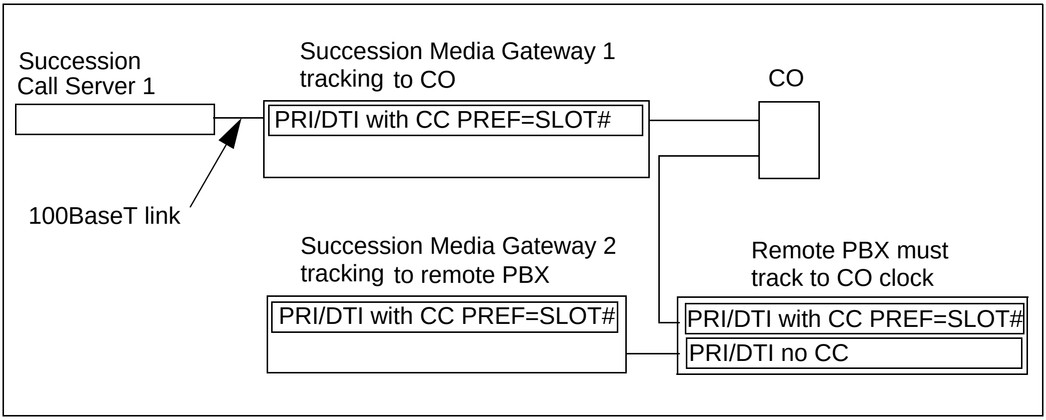


Figure 8
Unacceptable connection:
Succession Media Gateway 1 references CO; IP link to Succession Media Gateway 2 provides clocking

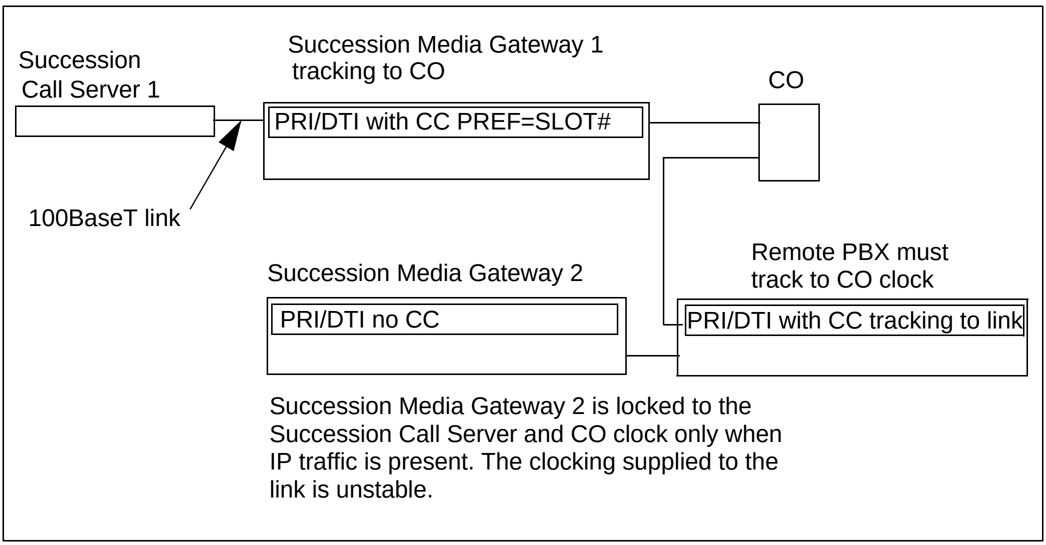
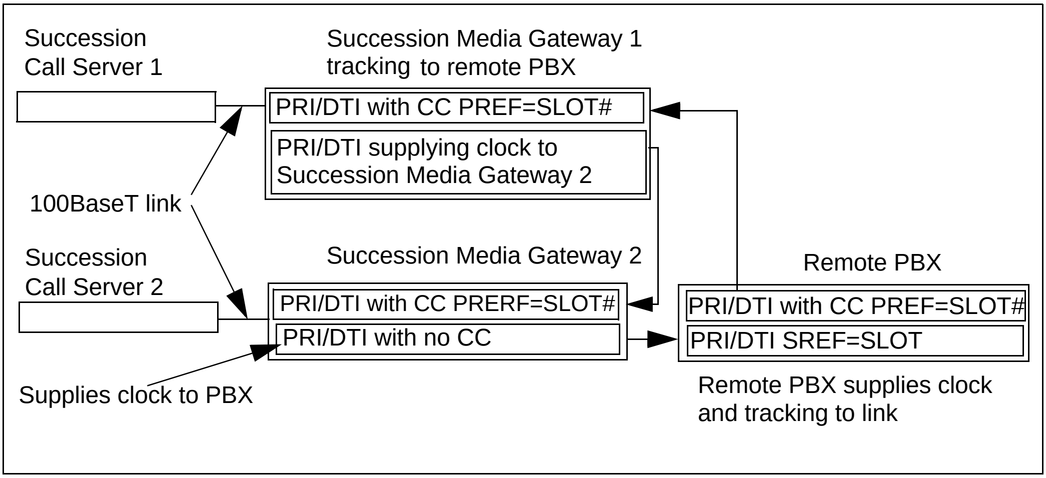


Figure 9
Unacceptable connection:
Succession Media Gateway 1 references remote PBX; Succession Media Gateway 2 provides master reference to remote PBX



Allocating primary and secondary references

The secondary reference (SREF) clock must reside in the main chassis with the primary (PREF) reference.

Figure 10
Acceptable connection:
Succession Media Gateway 1 references remote PBX; Succession Media Gateway 2 provides master reference to remote PBX

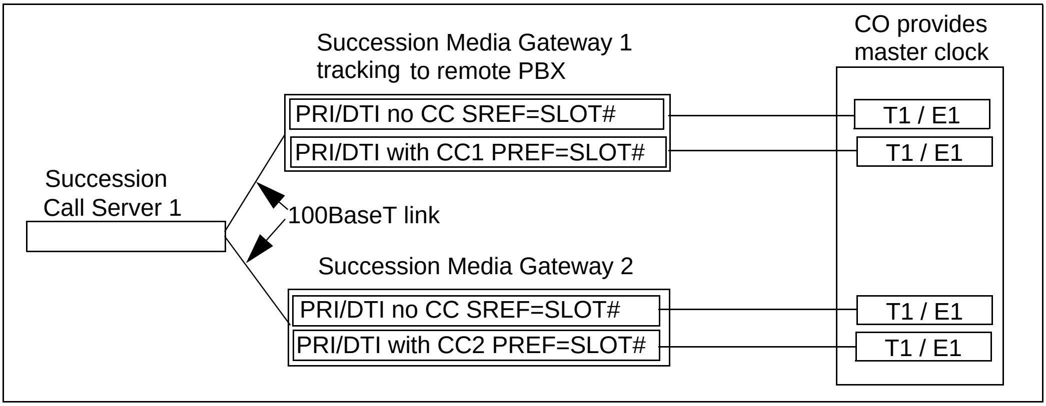
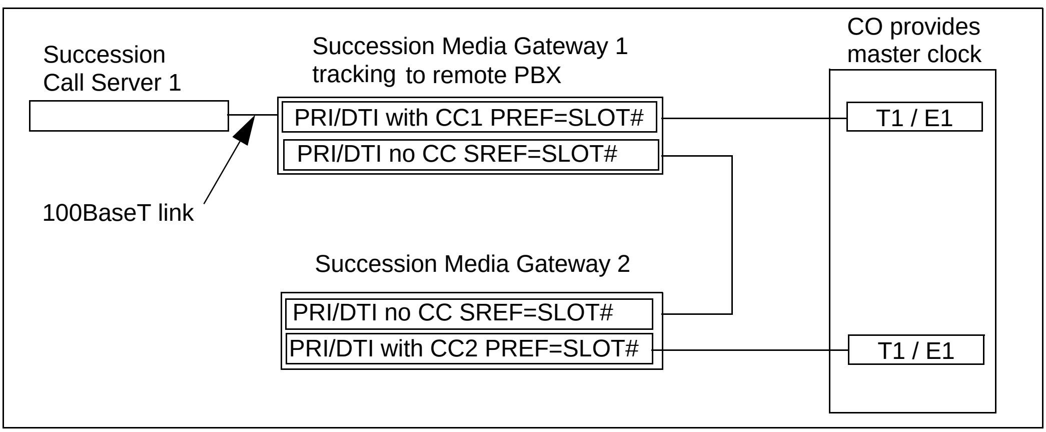


Figure 11
Acceptable connection:
Secondary reference provides backup reference when main link to CO is lost;
Secondary references can cross-reference to one another



Installation and configuration

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

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

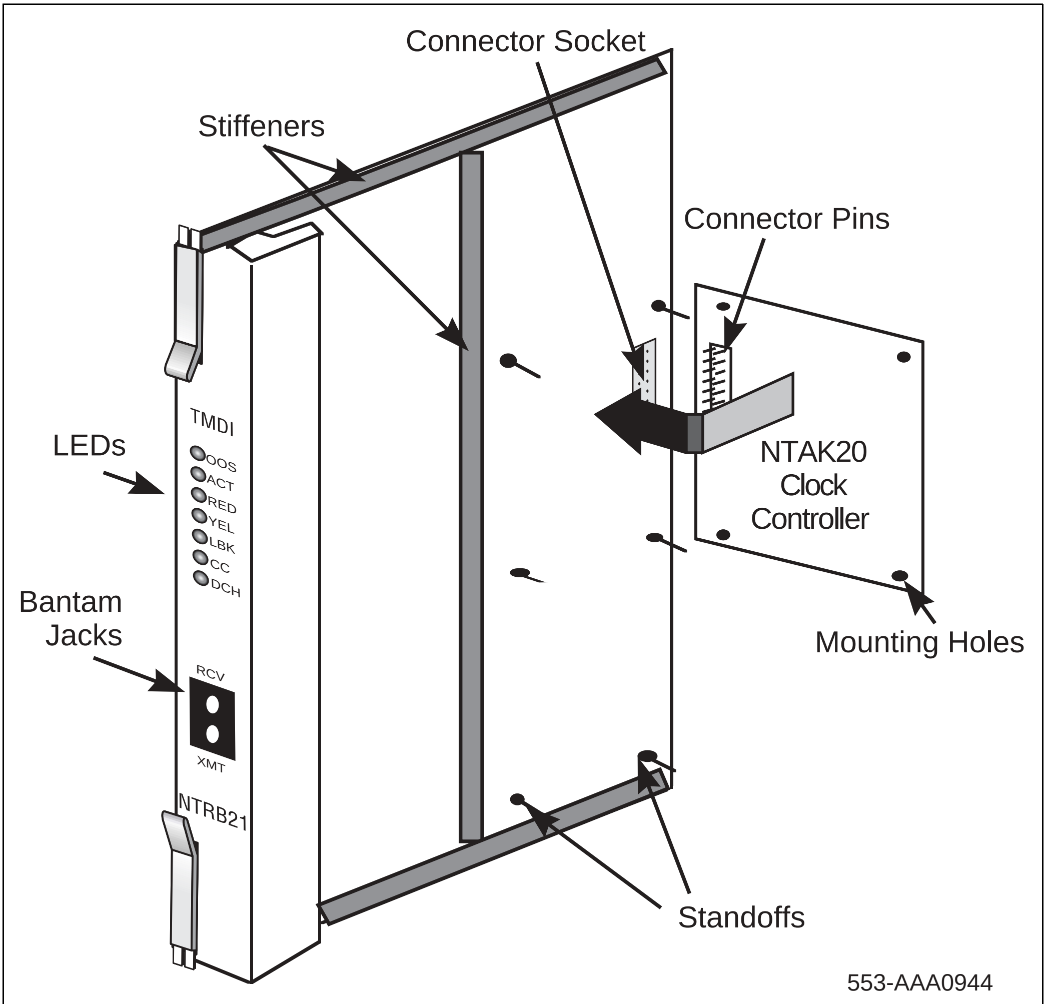
- NTRB21 – 1.5Mb DTI/PRI
- NTAK09 – 1.5Mb DTI/PRI
- NTBK22 – BRI
- NTBK50 – 2Mb PRI and MSDL

Embedded clock controllers are found on the following cards:

- NTAK10 – 2Mb DTI
- NTAK79 – 2Mb PRI

Figure 12 on page 48 shows the installation of the NTAK20 on the NTRB21 TMDI card.

Figure 12
NTAK20 Daughterboard installation



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

LD 73 - Configure clock controllers (Part 1 of 2)

Prompt	Response	Description
REQ	aaa	Request (aaa = CHG, END, NEW, OUT, or PRT)
TYPE	aaaa	Type (aaaa = DTI2, PRI2, or JDMI)
FEAT	SYTI	System Timers and counter (only one set per system) Valid response when TYPE = DTI2, JDMI, or PRI2
...		
CC1	xx	Card number for Clock Controller 1
PREF CC1	xx	Card number of DTI2/PRI2/SILC containing the primary clock reference. Where xx is 11-14 for Succession Media Gateway 1.
SREF CC1	xx	Card number of DTI2/PRI2/SILC containing the secondary clock reference. Where xx is 11-14 for Succession Media Gateway 1.
CC2	xx	Card number for Clock Controller 2
PREF CC2	xx	Card number of DTI2/PRI2/SILC containing the primary clock reference. Where xx is 21-24 for Succession Media Gateway 2.
SREF CC2	xx	Card number of DTI2/PRI2/SILC containing the secondary clock reference. Where xx is 21-24 for Succession Media Gateway 2.

LD 73 - Configure clock controllers (Part 2 of 2)

Prompt	Response	Description
CC3	xx	Card number for Clock Controller 3
PREF CC3	xx	Card number of DTI2/PRI2/SILC containing the primary clock reference. Where xx is 31-34 for Succession Media Gateway 3.
SREF CC3	xx	Card number of DTI2/PRI2/SILC containing the secondary clock reference. Where xx is 31-34 for Succession Media Gateway 3.
CC4	xx	Card number for Clock Controller 4
PREF CC4	xx	Card number of DTI2/PRI2/SILC containing the primary clock reference. Where xx is 41-44 for Succession Media Gateway 4.
SREF CC4	xx	Card number of DTI2/PRI2/SILC containing the secondary clock reference. Where xx is 41-44 for Succession Media Gateway 4.

Clock Controller commands (LD 60)

Command	Description
DIS CC n	Disable system clock controller n.
ENL CC n	Enable system clock controller n.
SSCK n	Get status of system clock n.
TRCK aaa n	Set clock controller tracking to primary, secondary or free-run. Where aaa is: • PCK = track primary clock • SCLK = track secondary clock • FRUN = free-run mode Track primary clock (PCK) or secondary clock (SCLK) as the reference clock or go to free-run (FRUN) mode.

Examples

Display the status of all Succession Media Gateway links.

SSCK 0 returns status all PLL links as follows:

```
DSBL->clock zero is not present in CSE systems

IP DB PORT 1  DSBL

IP DB PORT 2  LOCKING->Call Server is attempting to
lock to CAB 2

IP DB PORT 3  DSBL

IP DB PORT 4  DSBL

CABINET CLOCK SRC:  IP DB->The Call Server clock source
is from the IP DB
```

After a short period of time the status is as follows:

> ***Note:*** A voice or data call must be established in order to obtain LOCKED status unless the Zero Bandwidth parameter is set to NO in LD 117.

```
DSBL

IP DB PORT 1  DSBL

IP DB PORT 2  LOCKED->Call Server has locked to CAB 2

IP DB PORT 3  DSBL

IP DB PORT 4  DSBL

CABINET CLOCK SRC:  IP DB
```

Note: In a properly configured system you should not see the following in a print-out from the SSCK 0 command. Check the installed clock controllers and reconfigure as required.

```
DSBL

IP DB PORT 1  DSBLAll ports show DSBL. There is no
clock source to track to

IP DB PORT 2  DSBLand the system will have errors.

IP DB PORT 3  DSBL

IP DB PORT 4  DSBL

CABINET CLOCK SRC:  IP DB
```

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

Clock Controller commands (LD 60)

Command	Description
SSCK n	Get status of system clock n.

SSCK 1, 2, 3, or 4 returns the status of a configured clock controller.

```
.SSCK 1

ENBL

CLOCK ACTIVE

CLOCK CONTROLLER -  LOCKED TO SLOT 16

PREF - 16

SREF - 15

AUTO SWREF CLK - ENBL

IP DB PORT 2  DSBL

CABINET CLK SRC: CC
```

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

Clock Controller commands (LD 60)

Command	Description
TRCK PCK n	Set clock controller tracking to primary. Where n is 1, 2, 3, or 4: PCK = track primary clock Instructs the installed clock controller to track to a primary reference clock source. This is also referred to as “SLAVE” mode.
TRCK FRUN n	Set clock controller tracking to free-run. Where n is 1, 2, 3, or 4: FRUN = free-run mode Instructs the installed clock controller to free-run. In this mode, the system provides a reference or “MASTER” clock to all other systems connected through DTI/PRI links. This mode can be used only if there are no other clock controllers in SLAVE mode anywhere within the system.

The Succession Call Server can be locked to any Succession Media Gateway with the following command.

Clock Controller commands (LD 60)

Command	Description
TRCK PLL n	Overrides the default search order of cabinets 2, 1, 3, and 4. Where n is 1, 2, 3, or 4. Track primary clock (PCK) or secondary clock (SCLK) as the reference clock or go to free-run (FRUN) mode.

Example of the TRCK PLL command status SSCK 0:

```
. DSBL
IP DB PORT 1  LOCKED ->Call Server has locked to CAB 1
IP DB PORT 2  DSBL
IP DB PORT 3  DSBL
IP DB PORT 4  DSBL
CABINET CLOCK SRC:  IP DB
```

Data network planning for VoIP

Contents

This section contains information on the following topics:

Introduction

This chapter describes network requirements. These requirements are critical to the system Quality of Service (QoS).

Evaluating the existing data infrastructure

Organizations must 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 higher availability.

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

As you analyze your 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 is valuable in determining potential voice quality issues.

To evaluate availability requirements for the VoIP network, review network topology, feature capabilities, and protocol implementations. Also 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. You should evaluate the network baseline in terms of the impact on VoIP requirements.

To ensure that both IP Telephony and existing network requirements are met, you can add one or more of the following: memory, bandwidth, features.

Planning data networking for Succession 1000

To deploy the Succession 1000 system, consider the following data networking details and refer to *Data Networking for Voice over IP* (553-3001-160):

- 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/Peripherals
  - DNS
  - DHCP
  - TFTP
  - WEB server
  - QoS

QoS planning

An IP network must be engineered and provisioned to achieve high voice quality performance. The network administrator must implement QoS policies network-wide so 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 the following problems: speech breakup, speech clipping, pops and clicks, and echo. A best-effort network does not guarantee that bandwidth is available at any given time. End-users can use QoS mechanisms to ensure bandwidth is available at all times, and to maintain consistent, acceptable levels of loss, delay, and jitter.

For planning details for QoS, see *Data Networking for Voice over IP* (553-3001-160).

Core network planning

There are three networks:

1 Succession Call Server to Succession Media Gateway network

2 Management LAN (ELAN)

3 Voice LAN (TLAN)

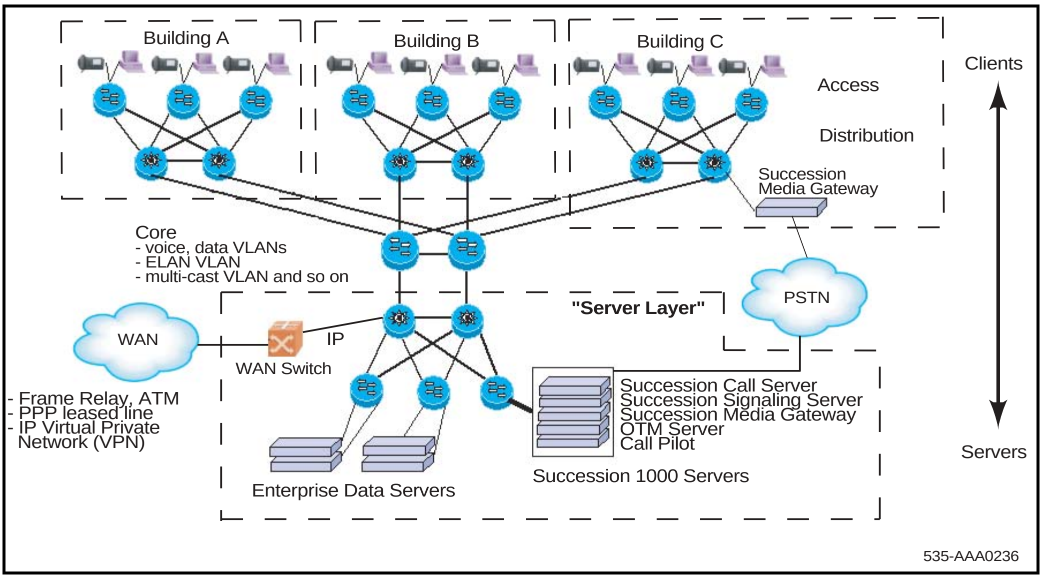
> ***Note:*** The Management LAN is also known as the ELAN. The ELAN (or Embedded LAN) isolates critical telephony signaling between the Succession Call Server and the other components. The Voice LAN is also known as the TLAN. The TLAN (or Telephony LAN) carries telephony/voice/signaling traffic, and connects to the customer network and the rest of the world.

100BaseTx IP connectivity

Between the Succession Call Server and Succession Media Gateways, the Succession 1000 supports 100BaseTx IP point-to-point connectivity or campus data network connectivity. Campus data network connectivity is provided through IP daughterboards in the Succession Call Server and the Succession Media Gateways. Figure 13 on page 61 shows the Succession Call Server and Succession Media Gateways connected over a large campus data network using 100BaseTx connectivity.

To satisfy voice quality requirements, adhere to applicable engineering guidelines. Refer to "Basic data network requirements for Succession Call Server to Succession Media Gateway connections" on page 63 and "Basic data network requirements for Internet Telephones" on page 68. Contact your local Data Administrator to obtain specific IP information.

Figure 13
Succession 1000 over a large campus data network

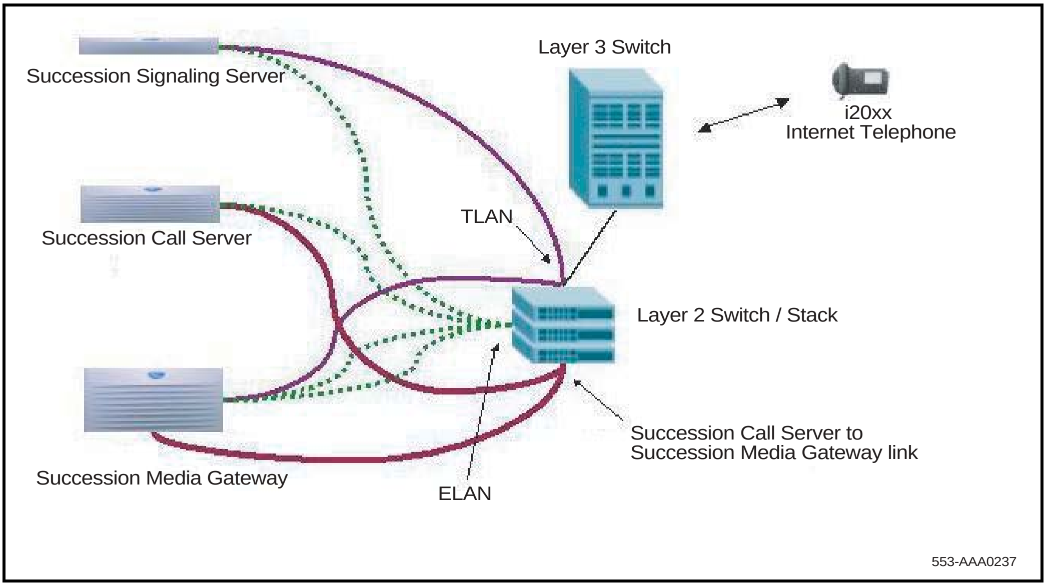


System network requirements

A typical Succession 1000 network configuration is shown in Figure 14 on page 62. The following network requirements are necessary:

- The ELAN and the TLAN must be on separate subnets.
- ELAN applications must be on the same subnet. This includes the Voice Gateway Media Cards (VGMC), which must be on the same ELAN subnet.
- VGMCs 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.

Figure 14
Succession 1000 network configuration



Basic data network requirements for Succession Call Server to Succession Media Gateway connections

Data network/LAN requirements

For excellent voice quality, the following requirements apply to the 100BaseTx connection between the Succession Call Server and the Succession Media Gateway.

- A 100BaseTx Layer 2 (or Layer 3) switch that supports full-duplex connection. Routers are not supported in Succession Call Server to Succession Media Gateway connections.

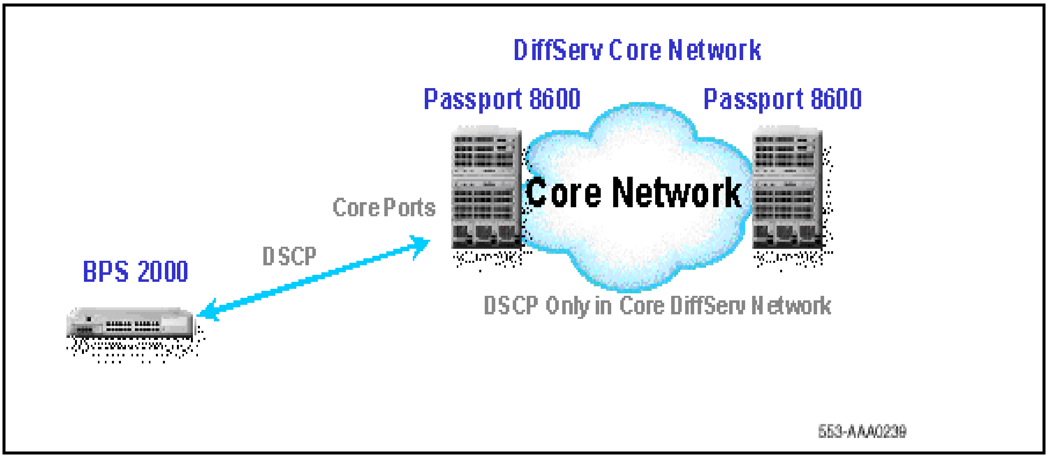
> **CAUTION**
> **Service Interruption**
>
> The ports on Layer 2 (or Layer 3) switching equipment must be set to auto-negotiate. When one side is hard configured, and the other side is auto-negotiate, the auto-negotiate side adapts to the fixed side speed settings, and sets itself to half-duplex. If the forced side is full-duplex, there is a duplex mismatch.

- Packet loss < 0.5%
- 100 Mbps IP link (minimum)
- Idle System Bandwidth: Approximately 0 Mbps
- Peak Bandwidth under high voice traffic (IP telephone to trunk calls) conditions: 10 Mbps
- Network delay is less then 5 msec Round Trip Delay (RTD) with PDV jitter buffer set to maximum
- Network delay is less then 12 msec Round Trip Delay (RTD) with PDV jitter buffer set to minimum
- Support of Port Priority Queuing: Recommended, but not required
- Support of VLAN configuration: Recommended, but not required

Auto-negotiate everything and hard-configure on a case-by-case basis. If this is not acceptable, then at least auto-negotiate LAN switch ports connected to Nortel Networks VoIP products. Auto-negotiate to concentrating devices (for example, to gateway cards and routers) and preferably, to telephones and 3-port switches. The IT support staff that manage LAN switches must be aware of the importance of this, and maintain tight controls on who can modify configurations.

Do not configure any true “hubs” in the critical path for VoIP. A LAN switch is a better choice, even for devices that only run 10m/half-duplex.

Figure 15
Basic LAN switch configuration for excellent voice quality



Packet Delay Variation (PDV) jitter buffer

Packet Delay Variation (PDV) jitter buffer smooths out variations in the arrival rate of the UDP/IP voice packets with respect to the rate at which the voice samples play. The minimum and maximum values for excellent voice quality are given in Table 12 on page 67. The PDV buffer also re-sequences out-of-order voice packets and is integral to the IP-based clock recovery scheme.

The PDV jitter buffer delay is adjustable and should be as short as possible. Insufficient delay is indicated when the QoS monitor reports buffer underflows, and causes a degradation in voice in the form of clicks or pops during a voice call. If you experience these problems, increase the size of the PDV buffer.

Increase the PDV buffer as little as possible to keep the round trip delay as short as possible. The goal is to operate with the smallest possible buffer to keep the round trip delay as short as possible.When increasing the buffer delay, increment in 0.5 msec steps until the QoS monitor no longer reports buffer underflows.

CAUTION
Loss of Data
Excessive delay causes a degradation in voice quality in the form of echo.

Bandwidth planning

The Succession 1000 system supports non-blocking transmission between the Succession Call Server and the Succession Media Gateways in an Internet Telephone application. With a mix of analog (500/2500-type) telephones, digital telephones, and Internet Telephones, the system can be engineered as blocking or non-blocking. The throughput of the network must be guaranteed.

Under high traffic conditions, a peak bandwidth of 10Mbps is consumed for voice traffic that requires Succession Media Gateway services (for example, trunk services).

> ***Note 1:*** A minimum 100 Mbps IP link is required.

> ***Note 2:*** If there is no traffic flow, there are no bandwidth requirements. Only active channels use bandwidth.

Table 11 on page 66 shows bandwidth consumptions over a 100BaseTx connection.

Table 11
Bandwidth consumption/100BaseTx

Talk Slot	Voice Traffic (Mbps)	Signaling Traffic (Mbps)	Total (Mbps)
100	9.7	0.5	13.8
75	7.8	0.5	10.2
40	5.6	0.5	6.1
16	4.1	0.5	4.6
0	0.0	0.11	0.11
Note: For voice traffic that requires Succession Media Gateway services.			

LAN recommendations for Excellent Voice Quality

Nortel Networks recommends that the Port-Based Virtual LAN (VLAN) feature be utilized to isolate the Succession 1000 system from the broadcast domain of the end-user's LAN equipment. This reduces the risk of link outages due to broadcast storms.

The packet prioritizing scheme can be used to effectively utilize bandwidth. However, network delay in a one-way trip delay must not exceed 2.5 ms. Support of priority queuing is recommended. Port priority queuing helps to maintain excellent voice quality during heavy usage or congestion.

Monitoring IP link voice quality of service

Behavioral characteristics of the network depend on factors like Round Trip Delay (RTD), Packet Delay Variation (PDV) jitter buffer, queuing delay in the intermediate nodes, packet loss, and available bandwidth. The service level of each IP link is measured and maintained on the Succession Call Server for the system. Information for latency and packet loss is collected from the hardware and processed.

Based on system configured thresholds, the service level is derived and reported with the **PRT QOS <cab#>** command in LD 117. See *Software Input/Output: Maintenance* (553-3001-511).

Data Network Ratings (Excellent, Good, Fair, Poor) along with the actual parameter values for network delay are displayed in Table 12.

Table 12
Campus data network voice quality measurements

Voice QoS Rating	Delay		Network Packet Loss
	Network Round Trip (PDV Max 7.8 ms)	**Network Round Trip (PDV Min 0.5 ms)**	
Excellent	< 5 ms	< 12 ms	< 0.5%
Good	5 - 25 ms	12 - 32 ms	0.5 - 1%
Fair	25 - 45 ms	32 - 52 ms	1 - 1.5 ms
Poor	> 45 ms	> 52 ms	> 1.5%

The values in Table 12 assume that there is no echo cancellation mechanism and no mechanism for recovering lost packets.

The command **PRT PDV <cab#>** in LD 117 displays the current size of the PDV buffer and the number of PDV underflows. In addition, a warning message is printed when a parameter threshold (or combination of thresholds) is reached. These thresholds are not user configured.

In LD 117, administrators can use the command **CHG PDV <port#> <delay>** to set Packet Delay Variation (PDV buffer size) on a per link basis. The **<delay>** parameter accepts values from 0.5 ms to 8 ms. This value should be initially tested at default settings. Increase the <delay> parameter value by 0.5 ms increments, if an unacceptable level of voice quality is experienced in the form of pops and clicks. Decrease this value if echo is experienced. The goal is to operate with the smallest buffer possible.

The PDV buffer size for each IP connection is configured at the Succession Call Server and automatically downloaded to the Succession Media Gateways.

Media conversion devices

Third-party media conversion devices can extend the range of the 100BaseTx and convert it to fiber. One such device, which has been tested by Nortel Networks, is the IMC Networks Ethernet Compatible Media Converter. It includes a McLIM Tx/Fx-SM/Plus module and provides acceptable transmission between the Succession Call Server and Succession Media Gateway if located up to 40 km apart. Use caution when extending the length of cable used in the point-to-point configuration — do not exceed the round trip delay parameters specified in Table 12 on page 67.

Basic data network requirements for Internet Telephones

For excellent voice quality, the following requirements apply to the 100BaseTx connection between the i2002 and i2004 Internet Telephone and the i2050 Software Phone, and the VGMC.

- End-to-End Delay, the sum of Encoding Delay + Jitter Buffer + Network Delay, should be: < 150 msec.
- Packet Loss (lost or late packets) for good quality: < 1%
  Tolerable Packet Loss (depending on codec): 3 to 5%

Bandwidth requirements

A dedicated 10BaseTx Ethernet segment at 35% load supports approximately:

- 37 simultaneous G.711, 20 msec/pkt conversations
- 170 simultaneous G.729AB, 50 msec/pkt conversations

A dedicated 100BaseTx Ethernet segment at 35% load supports approximately:

- 370 simultaneous G.711, 20 msec/pkt conversations
- 1,700 simultaneous G.729AB, 50 msec/pkt conversations

Thus, the recommended configuration is as follows:

- Connect each VGMC with a switched 10BaseTx Ethernet Connection.
- Share a 100BaseTx Ethernet Connection between every 10 VGMCs.

Bandwidth planning

- Partition the network into zones.
  - Most zones are characterized as a Region with one or more Ethernet LANs, possibly connected by a Wire Speed Router. Examples include: main office, remote office, and telecommuters, or each building in a campus.
  - Zones are inteconnected by slower speed WAN links, such as T1 Frame Relay.
- Select a codec for Intrazone use. For example, the G.711 with 20 msec packets.
- Verify that each zone has sufficient capacity.
  - Estimate traffic during busy hours.
  - Calculate the total bandwidth required (Busy hour calls x bandwidth), yielding, for example: 100 kbps for 20 msec G.711, full-duplex.
  - Budget 10 to 20% of network capacity for voice traffic on a shared voice and data network. For example, 10BaseTx switched hubs with 10BaseTx uplink network (Baystack 450) can support 100 to 200 calls on a shared network.
  - This configuration normally works. Do NOT recommend a large 10BaseTx unswitched network.
- Select a codec for Interzone use. For example, the G.729A with 40 msec packets, approximately 25 kbps full-duplex.
- Verify that there is sufficient capacity between each zone.
  - Estimate traffic (calls) leaving zone during busy hour.
  - Calculate the total bandwidth required (Busy hour calls x bandwidth).
  - Determine the current utilization of the WAN links for data during busy hour.
  - Add the voice and data bandwidth requirements.

- — Provision the Link with some Headroom. For example, 50% for Slow 64kbps link, or 20% for T1 Link.
- — If the Addition of Voice Traffic Exceeds Link Capacity plus Headroom, increase the Link.

Power requirements for i2004 and i2002 Internet Telephones

The i2004 and i2002 Internet Telephones require 16 V ac, 500 mA supplied by a local transformer. The appropriate transformer depends on the line voltage, which differs for each country.

- The NTEX00BA ships with a 117/120 V ac transformer for North America.
- The NTEX00BB does not include a transformer. Order the transformer appropriate to the country separately.

The i2004 and i2002 Internet Telephones also accommodate 48 V dc power. Power is applied by a “barrel” connector.

Site requirements

Contents

This section contains information on the following topics:

Introduction

Before installing the Succession 1000 system, the site must meet all environmental, grounding, power, and cross-connect terminal requirements.

Fire protection and safety requirements

Building, fire, and safety codes establish the degree of protection required for an installation. Additional information is available from the National Fire Protection Association (NFPA) in the following:

- “Standard for the Protection of Electronic Computer/Data Processing Equipment” (NFPA 75)
- “National Electrical Code (NEC)” (NFPA 70)

Fire protection and prevention

Expertise is required to properly locate and install sprinkler heads, fire and smoke sensing devices, and other fire extinguishing equipment. During the planning stage, consult experts, local codes, insurance underwriters, and local building authorities.

You can implement some fire precautions when an equipment area is constructed. For example, extend walls from floor to ceiling, and use non-combustible materials to construct walls, floors, and dropped ceilings.

If the structural floor is made of combustible materials, cover it with a non-combustible covering and remove all debris between the raised and permanent floors before system installation. If power connections reside beneath a raised floor, use waterproof electrical receptacles and connectors.

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

On a regular, scheduled basis, check services such as steam, water, and power, and inspect pipes for excess condensation, leaks, or corrosion.

Fire extinguishing systems

In most cases, carbon dioxide or water sprinkler systems are the recommended fire extinguishing systems. Dry-pipe water sprinklers are strongly recommended, because they interrupt power to the room and open a master valve that fills overhead sprinklers.

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

WARNING
Nortel Networks does not recommend using Halon or any other fire extinguishing system that is not described above. Nortel Networks supports Environmental Protection Agency restrictions on the use of other fire extinguishing systems.

Security precautions

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

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

Protect critical data, such as business records, by storing backups away from the equipment room. Nortel Networks highly recommends a regular updating program.

Safety procedures and training

Train company personnel to learn appropriate emergency response procedures. Some companies designate trained individuals as security members. Training should cover:

- when and how to evacuate personnel and records
- fire department notification
- electrical power shut off
- proper handling of fire extinguishers

Additionally, companies can install temperature and humidity monitoring devices (both visual and audible alarm signals) in equipment and storage rooms.

Occupational noise exposure

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

Environmental requirements

The environment in which the Succession 1000 system operates must meet the following general conditions:

- The room must be clean, relatively dust-free, and well ventilated. On equipment, ventilating openings must be free of obstructions.
- A Succession Call Server can dissipate up to 40 Watts of power.
- Each Succession Signaling Server can dissipate up to 125 Watts of power.
- Each Succession Media Gateway and Succession Media Gateway Expansion can dissipate up to 370 Watts of power. There must be enough ventilation in the equipment room to maintain an acceptable temperature.

- For an installed Succession Call Server, Succession Media Gateway, Succession Media Gateway Expansion, and Succession Signaling Server, maintain temperature between 0° and 45° C (32° and 113° F).
- Maintain humidity between 5% and 95% non-condensing.
- Select a location for equipment installation that is not subject to constant vibration.
- Locate equipment at least 12 ft (3660 mm) away from sources of electrostatic, electromagnetic, or radio frequency interference. These sources can include:
  - power tools
  - appliances (such as vacuum cleaners)
  - office business machines (such as copying machines)
  - elevators
  - air conditioners and large fans
  - radio and TV transmitters
  - high-frequency security devices
  - all electric motors
  - electrical transformers

Other environmental factors

In addition to temperature and humidity, control the following environmental factors in equipment areas:

- static electricity
- vibration
- electromagnetic and radio frequency interference (EMI/RFI)
- dust
- lighting
- structural features

Static electricity

Electronic circuits are highly sensitive to static discharge that can damage circuitry permanently, interrupt system operation, and cause lost data. Physical vibration, friction, and the separation of materials can cause static electricity. Other common causes include low humidity, certain types of carpeting, the wax on equipment room floors, and plastic-soled shoes. The human body is the most common collector of static electricity. A combination of plastic-soled shoes, certain flooring materials, and low humidity can cause body charges in excess of 15 kV.

Antistatic wrist straps, sprays, and mats are readily available. Nortel Networks recommends the use of an antistatic wrist strap whenever you work on Succession 1000 equipment.

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

Vibration

Vibration can slowly deteriorate mechanical parts and, if severe enough, result in serious disk errors. Avoid structure-borne vibration and consequent noise transferred to the equipment room. Raised floors require extra support jacks at strategic places to prevent the transmission of vibration.

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

Electromagnetic and radio frequency interference

Sources of EMI/RFI located close to system equipment can cause problems with system operation. The following are common EMI/RFI sources known to disturb system operation:

- thunderstorms, static electricity, and high-voltage power lines
- radar, broadcast stations, and mobile communications

- power tools, appliances (such as vacuum cleaners), and office business machines (such as copiers)
- industrial machines and ultrasonic cleaners
- vehicle ignition, arc welders, and dielectric heaters
- dimmer switches

Dust

Accumulated dust and dirt can degrade system reliability and performance by doing the following:

- scratching the contacts on circuit cards, causing intermittent failures
- having conductive contents that increase static electricity in the environment
- causing components to operate at higher temperatures

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

Lighting

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

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

Structural features

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

Selecting a site

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

- **Space**. The site must provide adequate space for unpacking, installation, operation, potential expansion, service, and storage. The site must provide space for sufficient cooling. Additional space may also be needed for a maintenance and technician area.
- **Location**. The location should be convenient for equipment delivery and close to related work areas. Consider the location of related equipment (such as the distribution frame and batteries) and the cable limitations.
- **Grounding and power**. Proper grounding and sufficient power facilities must be available.
- **Structural integrity**. The floor must support anticipated loads and, if applicable, the ceiling must support overhead cable racks.

Developing the site

Consider the following factors during site development:

- space and equipment layout requirements
- detailed floor plan and floor loading requirements
- building cable plan

The equipment room

Space and equipment layout requirements differ with each installation. Consider primary storage, secondary storage, and maintenance and technician space requirements when you plan the site.

Primary storage

The floor area required for a Succession 1000 system depends on the number of 19-inch racks required, the length-to-width ratio of the area, and the location of walls, partitions, windows, and doors. To determine the exact layout required, prepare a detailed floor plan after regarding all of the requirements in this chapter.

Although operating needs determine the general location of terminal devices, these devices must not be located beyond the maximum distances defined for their interface cards. Provide wall jacks and outlets for all devices in the equipment room.

Secondary storage

Provide space in the equipment area for storing disks, printer paper, printouts, and daily reports. A secure storage room for spare parts is recommended. Whenever possible, maintain the same environmental conditions in the equipment room and storage areas. If it is not possible to maintain the environment of the storage area exactly the same as the environment of the operating equipment, allow stored materials time to adjust to the equipment room environment before using them.

Maintenance and technician area

You can use the maintenance and technician area as an online work center and a place to store tools, test equipment, system documents, and spare parts. The area should have good lighting and convenient access to the Succession 1000 system. Typical items in a maintenance and technician area are as follows:

- shelves for instruction books
- spare parts storage room
- paper storage area
- locking cabinet or storage area for backup disks
- table or desk
- terminal, printer, or equivalent device

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

The floor plan

Prepare a detailed floor plan for each site, indicating the size and location of the following:

- system columns and modules, including planned expansion areas
- main distribution frame (MDF)
- service panel
- system terminal, printer, or other terminal devices (such as modems)
- external power equipment (such as rectifiers)
- cable racks
- Power Failure Transfer Units (PTFUs) and auxiliary power supplies (if either are equipped)
- space for additional equipment, such as reserve power equipment or auxiliary processors

System VoIP networking requirements

For detailed information on VoIP network requirements, refer to:

- *Data Networking for Voice over IP* (553-3001-160)
- *Succession 1000 System: Overview* (553-3031-010)

The network requirements are critical to the Succession 1000 system's quality of service. Ensure the following network requirements have been met:

- Provision 100BaseTx IP connectivity between the Succession Call Server and the Succession Media Gateways. The 100BaseTx IP connectivity can be either a point-to-point network or a distributed campus data network. IP daughterboards in the Succession Call Server and the Succession Media Gateways provide connectivity.
- Ensure that the 100BaseTx Layer 2 (or Layer 3) switch supports full-duplex connection. Routers are not supported in Succession Call Server to Succession Media Gateway connections.

  Note: The ports on Layer 2 (or Layer 3) switching equipment must be set to auto-negotiate ENABLED.

- Provision the ELAN and the TLAN on separate subnets.
- Provision all applications on the ELAN on the same subnet. This includes VGMCs that must be on the same ELAN subnet.
- VGMCs in the same node must be on the same TLAN subnet.

Basic data network requirements for Internet Telephones

For excellent voice quality, ensure that the 100BaseTx connection between the Internet Telephones and the VGMCs must meet the guidelines specified in the *Succession 1000 System: Overview* (553-3031-010).

Power requirements for Internet Telephones

The i2004 and i2002 Internet Telephones require a 16 V ac, 500 mA supplied by a local transformer. The appropriate transformer depends on the line voltage, which is different for each country.

- The NTEX00BA ships with a 117/120 V ac transformer for North America.
- The NTEX00BB does not include a transformer. Order the transformer appropriate to the country separately.

The i2004 and i2002 Internet Telephones also accommodate 48 V dc power, applied by a “barrel” connector.

The building cable plan for circuit-switched equipment

In the building cable plan, show the routing of all wiring, the location and wiring requirements for each terminal device connected to the system, and any other relevant information about the device. Also, show the location of distribution frames, conduits, access points, and power outlets.

In addition, perform the following tasks:

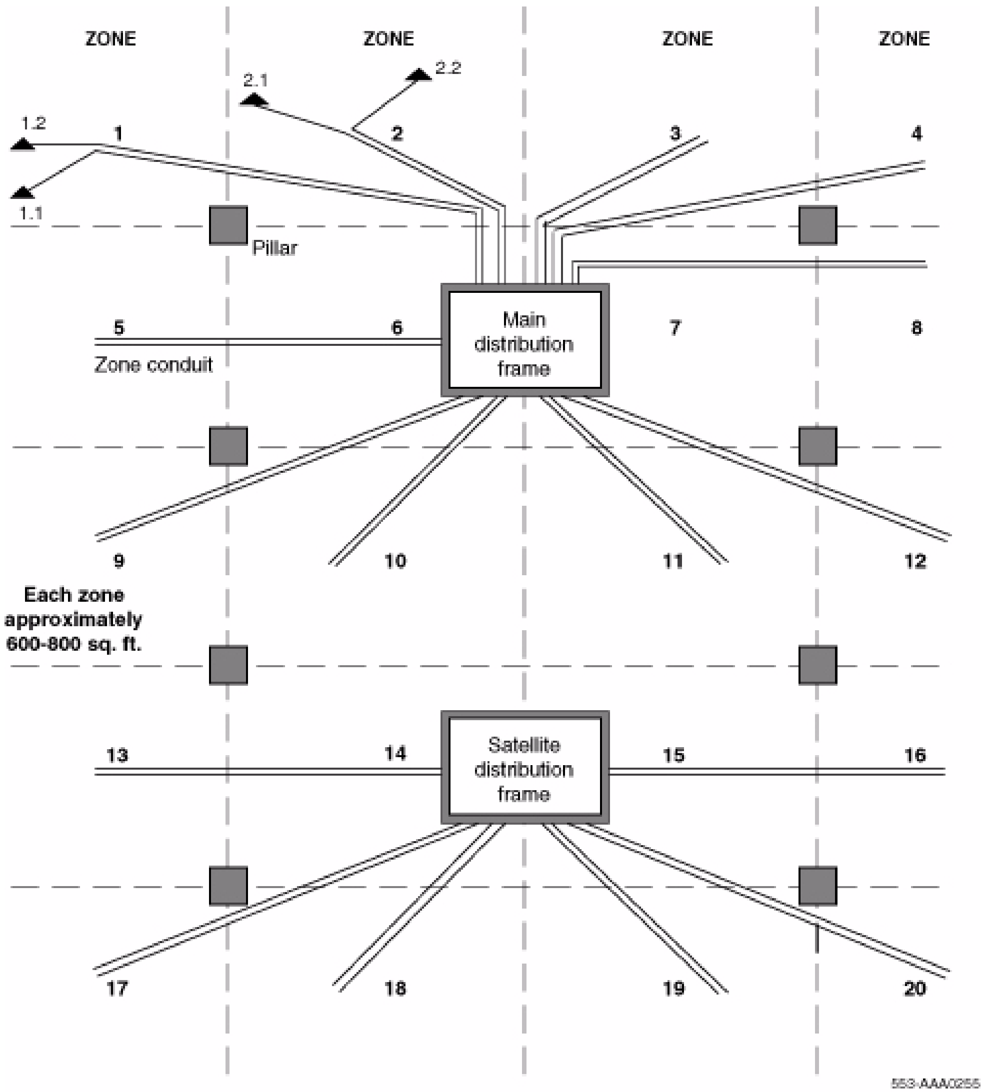
- Identify ownership of existing building wire, if using any.
- Perform a random sampling of in-place wiring to ensure that it meets specifications for high-speed lines. All wiring carrying high-speed data must pass a verification test as part of the installation process.

- Identify the location of conduits and floor ducts. When telephone cable runs in conduit, that conduit must not be used for any other wiring.
- Identify the location of all distribution points, main and intermediate.
- Provide three pairs of telephone wire from a distribution frame to a nearby telephone jack for each terminal device. Modular jacks must be within 2.4 m (8 ft) of the device.
- Provide a 16-pair (or 25-pair) cable equipped with an Amphenol-type connector for each attendant console.

Divide the building cable plan into zones. Zones are typically the termination point of conduits throughout the office. Identify each zone on the building cable plan with a letter or number, and assign a block of numbers to each zone. Figure 16 on page 83 provides an illustration of zoning.

Note: Be sure to leave room for expansion.

Figure 16
Building cable zones



Wire routing

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

- **Floors**. In the open, wires can run along baseboard, ceiling moldings, or door and window casings. For the safety of employees, never run wire across the top of the floor. When concealed, wires can run inside floor conduits that travel between distribution frames and jacks.

  Note: Under-carpet cable is not recommended.

- **Ceilings**. National and local building codes specify the types of telephone wire that can run in each type of ceiling. Local building codes take precedence.
- **Walls**. Cables that run vertically should, when possible, run inside a wall, pole, or similar facility for vertical wire drops. Cables that run horizontally cannot be blind-fed through walls.
- **Between floors**. Locate distribution frames as closely to one another as possible. Local coding laws specify whether or not a licensed contractor is required if a conduit is installed.
- **EMI**. Data degradation can occur if wires travel near strong EMI sources. See “Environmental requirements” on page 74 for a description of common sources of interference.

Termination points

Once you determine the wire routing, establish termination points. Cables can terminate at the following locations:

- the MDF (typically in the equipment room)
- intermediate distribution frames (typically on each floor in telephone utility closets)
- wall jacks to terminal boxes (typically located near the terminal device)

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

In all cases, clearly designate the block where the cables terminate with the cable location information and the cable pair assignments. Keep a log book (cable record) of a termination information.

See Figure 17 on page 86 for an example of a cable record.

Figure 17
Sample cable record

CABLE RECORD

Customer ______________________________________

Location ____

Cable ______________________ Binder ____________ Page ____ of ____

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

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Preparing for delivery

Consider the following questions when you plan for delivery:

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

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

Preinstallation inspections

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

- Inspect the equipment room to verify that:
  - all physical and environmental requirements are met
  - system grounding and power equipment is installed and tested
  - the equipment layout is marked on the floor
- Inspect building cable to verify that:
  - sufficient distribution frames are provided
  - conduits or floor ducts to terminal locations are installed
  - terminal jacks are installed
  - sufficient wiring is on hand

The delivery route

Before the Succession 1000 system is delivered, examine and measure the route from the receiving area to the installation area. Consider the following factors:

- size and security of unloading and storage areas
- availability and capacity of elevators
- number and size of aisles and doors on the route
- restrictions, such as bends or obstructions, in halls or at doors
- floor loading capacity of unloading, storage, and equipment room areas
- number of steps and stairways

Preparing for installation

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

The installation plan

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

The equipment room floor plan should show the following:

- location of the system 19-inch rack, including planned expansion areas
- main distribution frame
- service panel
- system terminal, printer, or other terminal devices
- external power equipment (such as rectifiers)
- cable racks
- PFTUs and auxiliary power supplies (if either are equipped)
- additional equipment, such as reserve power equipment or auxiliary processors

The building cable plan should show the following:

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

An Installation and Test Sequence (ITS) chart shows typical installation tasks, task sequence, and task start and duration information.

Work orders

The work order can include the following:

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

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 or problems are encountered.

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

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

Before installing the Succession 1000 equipment and applying ac power, measure the impedance of the building ground reference. An ECOS 1023 POW-R-MATE or similar meter, is acceptable for this purpose. Ensure that the ground path connected to the Succession 1000 system has an impedance of 4 ohms or less. Make any improvements to the grounding system before you install the Succession 1000 system.

CAUTION
Never connect the single point ground conductor from the Succession 1000 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.

Additional grounding requirements are as follows:

- Ground conductors for the Succession 1000 system must not:
  - be smaller than #6 AWG (#40 metric) at any point (see Table 13 on page 91 for a list of grounding wire requirements specific to some areas)
  - carry current under normal operating conditions
- 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 13
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 two 10 mm^2

CAUTION
For an installed Succession Call Server, Succession Media Gateway, Succession Media Gateway Expansion, or Succession Signaling Server, link impedance between the ground post of any equipment and the single point ground to which it connects must be less than 0.25 ohms.

CAUTION
Transients in supply conductors and ground systems can damage integrated circuits. This damage can result in unreliable Succession 1000 operation. Damage caused by transients is not always immediately apparent. Degradation can occur over a period of time.

Single Point Grounding

Correct grounding of communications systems is necessary to protect equipment from the hazards of surge and noise interference. The Single Point Grounding (SPG) method of protecting communications equipment is the Nortel Networks standard.

The requirements for Single Point Grounding are described in the following sections: Safety, Protection, EMC, Installation and Maintenance, Powering, and Advances in Technology.

Safety

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

CAUTION
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.

Protection

Correct grounding is necessary to protect equipment. This includes grounding for outside plant cable shields and protectors, and the grounds for framework, battery, and logic references.

EMC

To ensure adequate emission and susceptibility performance, equipment must conform with Electromagnetic Compatibility (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 end-user's premises. Correcting violations of national codes after the initial installation is difficult and costly.

Powering

Grounding system design must consider the power options for the equipment. If the equipment is backed up with an Uninterruptible Power Supply (UPS), the design must 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

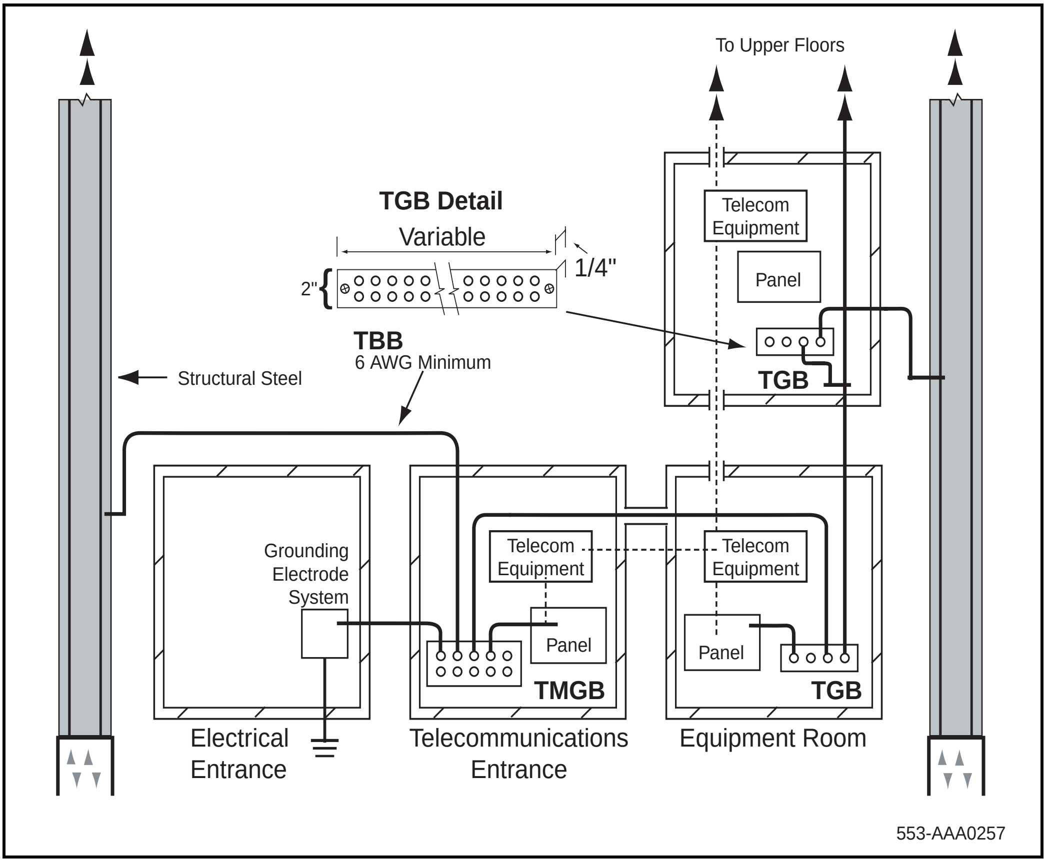
The component density on circuit cards continues to increase because of the miniaturization and multi-layering of printed circuit boards. The operating speeds of the integrated circuits are also increasing. Grounding provides important protection for these components.

In a system, the Telecommunications Main Grounding Busbar (TMGB) / Telecommunications Grounding Busbar (TGB) is the point at which telecommunications equipment bonds to the ground. A copper busbar normally acts as the system SPG. You can use any of the following busbars 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 TMGB/TGB bonded to the building grounding system

Configure telecommunications subsystems, such as groups of frames or equipment, as separate single point ground entities connected to the building's SPG using the TMGB and the TGBs. Refer to Figure 18 on page 94.

Figure 18
ANSI/TIA/EIA-607 Grounding Schematic



Grounding method

The grounding method used for the Succession 1000 equipment depends on whether the same service panel powers all systems. The following three grounding scenarios are possible:

1 A system with one Succession Media Gateway.

2 A system with more than one Succession Media Gateway, powered by the same service panel.

3 A system with more than one Succession Media Gateway, powered by multiple service panels.

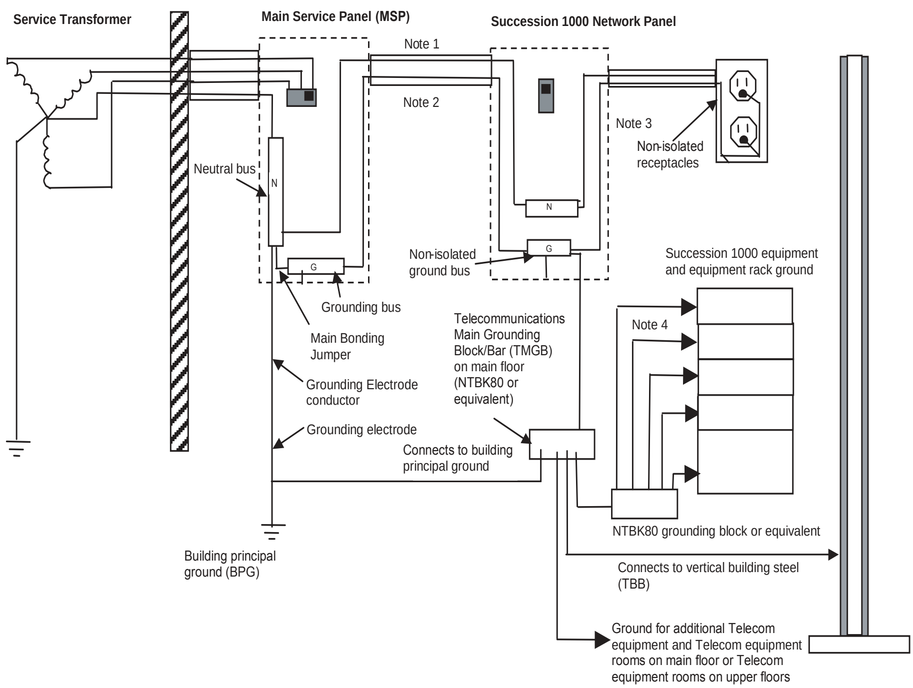
A system with one Succession Media Gateway or multiple Succession Media Gateways powered by one service panel

For each Succession 1000 component, connect a #6 AWG (#40 Metric Wire Gauge) ground wire from the rear panel grounding lug to the NTBK80 grounding block. See Table 13 on page 91 for specific grounding wire requirements in some areas. Connect the grounding block to a ground source (TMGB or TGB). Consider the Succession Media Gateway and the Succession Media Gateway Expansion as the same ground. Jumper the ground wire from the Succession Media Gateway Expansion to the Succession Media Gateway and then back to the grounding block.

CAUTION
To prevent ground loops, power all Succession 1000 equipment from the same dedicated power panel.

- Ground all Succession Call Servers, Media Gateways and Succession Signaling Servers to the power panel through the grounding block.
- Ground the Succession Media Gateway Expansion to the Succession Media Gateway.

Figure 19
Typical wiring plan



Notes:

1 Panel phase conductors are not shown for drawing clarity.

2 Nortel Networks recommends that the system ground conductor be insulated and the same size as the largest conductor run between the Main Service Panel (MSP) and Succession 1000 Network Panel.

3 Dedicated a/c service panel must be located in the equipment room as close to the Succession 1000 system as practically possible.

4 Conductors must not be smaller than #6 AWG.

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Equipment powered by multiple service panels

For each Succession Media Gateway, connect a #6 AWG (#40 Metric Wire Gauge) ground wire from the equipment rear panel grounding lug to the NTBK80 grounding block. See Table 13 on page 91 for grounding wire requirements specific to some areas. If any system cannot be powered from the same service panel, ground it to the nearest TGB point. Power each Succession Media Gateway and Succession Media Gateway Expansion pair from the same service panel.

> ***Note:*** In the UK, connect the grounding wire from the Succession 1000 equipment to an NTBK80 grounding block or through a Krone Test Jack Frame.

Grounding multiple components in a rack

Ground each component in a rack. If a component does not have a ground lug, ground the whole rack.

Conduit requirements

Conductive conduit linking panels and equipment are legal for use as a grounding network in most countries. For all system ground paths for the Succession 1000 system, use the correct size of insulated copper conductors routed inside conduit. A ground link that depends on a conduit can defeat improvements made by installing dedicated panels and transformers. The following reasons explain why:

- Personnel who service different equipment can separate conduit links. If such a separation occurs between the Succession 1000 system and the building ground reference, the conduit cannot provide a ground path. This situation is hazardous.

- Metal conduits often corrode, especially at threaded connections. Corrosion increases resistance. This problem becomes worse when multiple links are involved. Applying paint over the conduit increases the corrosion process.
- Always fasten the conduit to secure surfaces. Often, the conduit bolts on to structural steel members, which can function as ground conductors to noisy equipment (for example, compressors and motors). Adding noisy equipment into the Succession 1000 grounding system can damage its performance, and resulting intermittent malfunctions can be difficult to trace.

Commercial power requirements

The Succession 1000 system is available with ac power only. The optimal installation of the ac-powered Succession 1000 system includes a direct connection to the electrical system in the building, provided some requirements are met. Refer to ac-powered installation on page 105 for detailed information.

The Succession Media Gateway and Succession Media Gateway Expansion can share the same electrical breaker and outlet in accordance with the local electrical codes.

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” on page 99. Refer to “Power consumption worksheets” on page 111 to determine the power consumption of the Succession 1000 system.

WARNING

The circuit breaker loading requirements are intended to provide optimal reliability in case of an individual system component power failure (which causes the breaker to trip). Specific circuit loading requirements can be calculated and implemented using Tables 15 through 23, but system reliability may be sacrificed.

Alternative ac-powered installation

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

- 120/208/240 V ac input, over-current protected at primary
- 120/208/240 V ac available at secondary outputs, each circuit breaker-protected
- Completely isolate 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 Succession 1000 components operating at the same time at full load
- Equipment unrelated to the Succession 1000 system must not be powered from a transformer that provides service to the Succession 1000 system

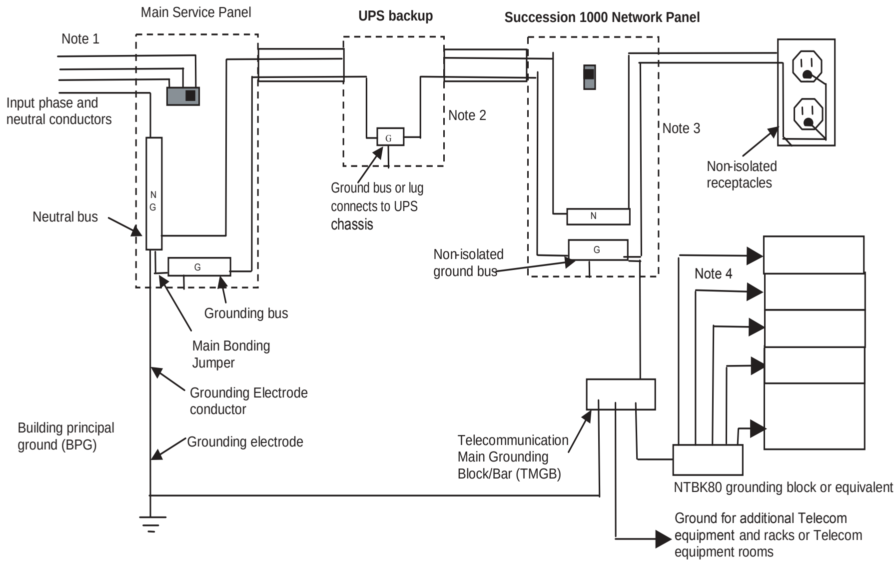
Uninterruptible Power Supply

For backup ac power, use an Uninterruptible Power Supply (UPS) to feed the Succession 1000 system. The power requirement for a UPS is 550 VA per system. Install the UPS according to the manufacturer's instructions. The maximum power requirement for Succession 1000 equipment on the same breaker is 1100 VA.

> ***Note:*** 1100 VA is an absolute maximum. To calculate the minimum requirements for your configuration, use Table 24, "Circuit card power consumption," on page 111.

Figure 20 on page 100 shows a typical UPS wiring plan.

Figure 20
Typical UPS wiring plan



Notes:

1 Panel phase conductors are not shown for drawing clarity.

2 Nortel Networks recommends that the system ground conductor be insulated and the same size as the largest conductor run between the power supply source and Succession 1000 Equipment Panel.

3 Dedicated a/c service panel must be located in the equipment room as close to the Succession 1000 system as practically possible.

4 Conductor must not be smaller than #6 AWG.

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Isolation transformer ground

The transformer ground must have separate grounds for primary and secondary windings, rather than a common ground.

Ground conductors inside the transformer must be correctly sized.

Procedure 1
Installing an isolation transformer with pluggable power cords

1. Connect the power cords of all Succession 1000 components to the outlets on the transformer secondary.

2. Attach a separate conductor between the lug on the transformer and the Main building or floor ground. Place a "DO NOT DISCONNECT" tag on the conductor.

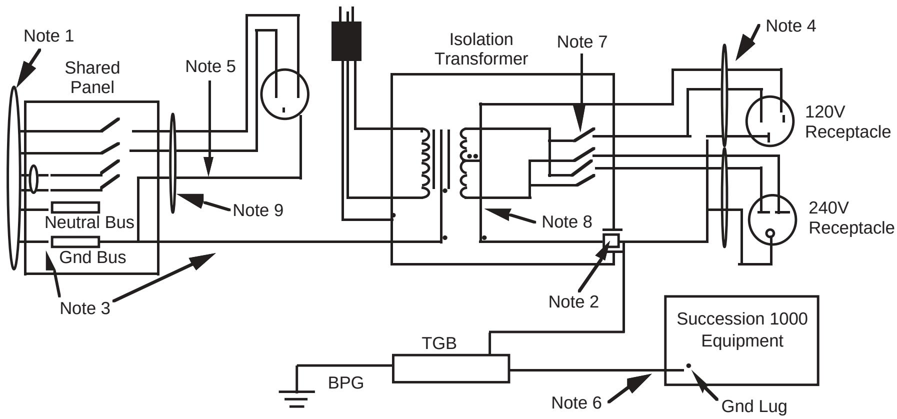
3. Fasten or tie this conductor to the TGB feeding the Succession 1000 system.

   Note: Power all equipment related with the Succession 1000 from the secondary of the transformer only. Ground all equipment to the TMGB or the TGB. Do not connect equipment to the isolation transformer that powers the Succession 1000 system if that equipment is not related to the Succession 1000 system.

4. Power the transformer primary through a dedicated circuit.

CAUTION
Nortel Networks does not recommend connecting any telecommunications ground bus of the Succession 1000 system to untested horizontal structural steel or water pipes, or other unreliable ground paths. Use a ground point known to be "clean" and permanent. Place a "DO NOT DISCONNECT" tag on it. Figure 21 on page 102 shows the pluggable cord connections.

Figure 21
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 connecting 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 Succession 1000 equipment SPG insulated ground conductor as close as possible to the main building ground reference. Isolate the ground bus from panel housing if permitted by local codes. Conductor should be a minimum of 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 can be made by metallic conduit. Additional copper conductor recommended.

6 Minimum AWG #6 (metric #40) insulated copper conductor connected to FGND lug on the chassis. Route separately from AC power cable.

7 Separate breaker required for each Succession Media Gateway. Breakers must be mounted on the transformer if the receptacles are also mounted on the transformer. If they are in a panel served by the transformer secondary, all connections between the receptacles and transformer must be in conduit. The Succession Media Gateway and the Succession Media Gateway Expansion can coexist on the same breaker.

8 Connect secondary neutral (X0) to system SPG.

553-AAA0141

End of Procedure

Procedure 2
Installing an isolation transformer without pluggable power cords

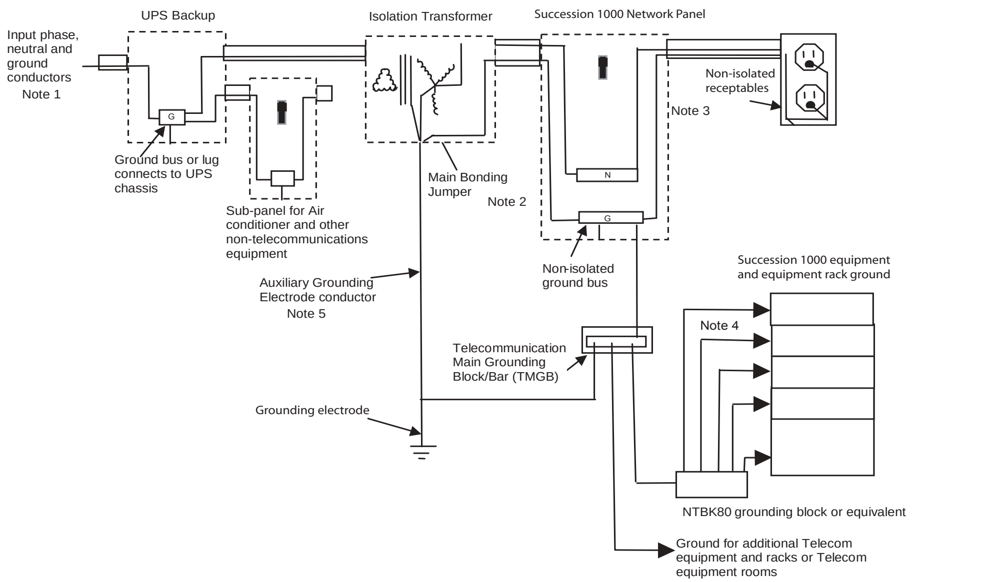
1 If the transformer does not have a pluggable cord, hardwire the transformer to an electrical panel. Route all wires (including grounds) through a single conduit.

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

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

3 Run all ground lines through the same conduit as the phase conductors that serve the equipment. Figure 22 on page 104 shows the isolation transformer connections.

Figure 22
Typical hardwired isolation transformer wiring plan



Notes:

1 Panel phase conductors are not shown for drawing clarity.

2 Nortel Networks recommends that the system ground conductor be insulated and the same size as the largest conductor run between the power source and Succession 1000 Panel.

3 Dedicated a/c service panel must be located in the equipment room as close to the Succession 1000 system as practically possible.

4 Conductor must not be smaller than #6 AWG.

5 Auxiliary grounding conductor should be short as possible, and must be connected to a recognized building ground. (That is, floor ground bar, vertical ground riser, or building principal ground.)

553-AAA0142

End of Procedure

The ac power installation for systems installed in a rack

If other data communications equipment is in the same rack as the Succession 1000 system, power each piece of equipment from the same service panel. Power from each outlet must meet the input requirements of at least one Succession 1000 power supply, as listed in Tables 15 through 23. Check power requirements for other system equipment. Install additional outlets, if necessary.

Table 14
Input power required by equipment and region

The ac input requirements for each...	North America	Europe and the UK	Germany
Succession Media Gateway or Succession Media Gateway Expansion	See Table 15 on page 106.	See Table 16 on page 106.	See Table 17 on page 107.
Succession Call Server	See Table 18 on page 107.	See Table 19 on page 108.	See Table 20 on page 108.
Succession Signaling Server	See Table 21 on page 109.	See Table 22 on page 109.	See Table 23 on page 110.

Table 15
The ac input requirements for each Succession Media Gateway or Succession Media Gateway Expansion
(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 Volts, 15 Amp supply

Table 16
The ac input requirements for each Succession Media Gateway or Succession Media Gateway Expansion
(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 Volts, 15 Amp supply
Note 1: Because local power specifications vary, consult 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 17
The ac input requirements for each Succession Media Gateway or Succession Media Gateway Expansion (Germany)

Voltage	Recommended: 230 Volts Maximum limits: 180 and 250 Volts Single phase
Frequency	50 Hz
Power (I/P max)	550 VA maximum
Fuse	16 A
Outlet Type	Receptacles by DIN regulation

Table 18
The ac input requirements for each Succession Call Server (North America)

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

Table 19
The ac input requirements for each Succession Call Server (Europe and UK)

Voltage	Recommended: 208/220 Volts Maximum limits: 180 and 250 Volts Single phase
Frequency	50-60 Hz
Power (I/P max)	60 VA maximum
Outlet Type	208/240 Volts, 15 Amp supply

Note 1: Because local power specifications vary, consult 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 20
The ac input requirements for each Succession Call Server (Germany)

Voltage	Recommended: 230 Volts Maximum limits: 180 and 250 Volts Single phase
Frequency	50 Hz
Power (I/P max)	60 VA maximum
Fuse	16 A
Outlet Type	Receptacles by DIN regulation

Table 21
The ac input requirements for each Succession Signaling Server (North America)

Voltage	Recommended: 100-120 Volts Maximum limits: 90 and 132 Volts Single phase
Frequency	50-60 Hz
Power (I/P max)	200 VA maximum
Outlet Type	120 Volts, 15 Amp supply

Table 22
The ac input requirements for each Succession Signaling Server (Europe and UK)

Voltage	Recommended: 208/220 Volts Maximum limits: 180 and 250 Volts Single phase
Frequency	50-60 Hz
Power (I/P max)	200 VA maximum
Outlet Type	208/240 Volts, 15 Amp supply
Note 1: Because local power specifications vary, consult 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 23
The ac input requirements for each Succession Signaling Server (Germany)

Voltage	Recommended: 230 Volts Maximum limits: 180 and 250 Volts Single phase
Frequency	50 Hz
Power (I/P max)	200 VA maximum
Fuse	16 A
Outlet Type	Receptacles by DIN regulation

Site requirements

A dedicated circuit breaker panel is recommended for an optimal ac-powered Succession 1000 system. If a dedicated panel cannot be provided, an isolation transformer may be required. Refer to "Alternative ac-powered installation" on page 99.

A dedicated circuit breaker panel provides power only to the Succession 1000 system and its related telecommunications hardware, such as TTYs and printers.

> ***Note:*** You cannot always power a complete system from a single circuit-breaker panel. For example, when components of the Succession 1000 are in remote locations.

Location of power outlets

The maximum distance between a power outlet and the system equipment depends on the length of the power cord.

- In North America, the power cord is 9 ft 10 in. (3000 mm).
- Outside North America, the power cord is 8 ft 2 in. (2490 mm).

Power consumption worksheets

Use the worksheets (Tables 25 to 29) in this section to determine the power consumption for the Succession 1000 system. Refer to Table 24 for the circuit card power consumption.

Table 24
Circuit card power consumption

Circuit card	Type	% active sets (off-hook)	Power consumption
NT5K02	Flexible analog line card	50%	26W
NT8D02	Digital line card	100%	25W
NT8D03	Analog line card	50%	26W
NT9D09	Message-waiting line card	50%	26W
NT8D14	Universal trunk card	DID-enabled	28W
NT8D15	E&M trunk card	N/A	29W
NTDK20	Succession System Controller Card	N/A	N/A
NTAK02	SDI/DCH card	N/A	10W
NTAK03	TDS/DTR card	N/A	8W
NTAK09	1.5Mb DTI/PRI card	N/A	10W
NTAK10	2.0 Mb DTI card	N/A	12W
NTAK79	2.0 Mb PRI card	N/A	12W
NTBK50	2.0 Mb PRI card	N/A	12W
NTV001	VGMCs	N/A	30W

Table 25
Succession 1000 power consumption: Succession Media Gateway

Slot	Circuit card	Type	Power consumption from Table 24
0	NTDK20	SSC	35W
1			
2			
3			
4, 5, 6			
		Total Power Out	

Table 26
Succession 1000 power consumption: Succession Media Gateway Expansion

Slot	Circuit card	Type	Power consumption from Table 24
7			
8			
9			
10			
		Total Power Out	

Table 27
Succession 1000 power consumption: Succession Call Server

Slot	Circuit card	Type	Power consumption from Table 18, Table 19, and Table 20
1	N/A	N/A	60W
		Total Power In	60W

Table 28
Succession 1000 power consumption: Succession Signaling Server

Slot	Circuit card	Type	Power consumption from Table 21, Table 22, and Table 23
1	N/A	N/A	200W
		Total Power In	200W

Table 29
Total Succession 1000 system power consumption

Succession Media Gateway Power Out from Table 25 on page 112 (Total for slots 0-4 in the Succession Media Gateway)	
Succession Media Gateway Expansion Power Out from Table 26 on page 112 (Total for slots 7-10 in the Succession Media Gateway Expansion)	
Total Out	
Total Out x 1.5 efficiency allowance = This is the total system ac Input Power requirement.	

Auxiliary equipment power

Terminals, printers, modems, and other data units used with the Succession 1000 system require special wiring considerations. Power for system equipment in the switch room must meet the following requirements:

- Switch room equipment must be powered from the same panel or transformer as the Succession 1000 system.
- Switch room equipment must be grounded to the same panel or transformer as the Succession 1000 system.
- Switch room equipment must be labeled at the panel to prevent interruption that is not authorized.
- Switch room equipment must not be controlled by a switch between the breaker and the equipment.

Service receptacles for Succession 1000 ac systems and related equipment must be the following:

- non-isolated ground type
- rated for 120 or 240 V, 15 or 20A, 50-60 Hz, 3-pole, 3-wire, grounded

Maintenance and administration equipment

Refer to *System Management* (553-3001-300) for information about communicating with Succession 1000 system. Refer to *Succession 1000 System: Installation and Configuration* (553-3031-210) for information about configuring modems and terminals.

To communicate with the Succession 1000 system through either a direct or remote connection, the supported devices are:

- Maintenance workstation
  - equipped with a dial-up modem or connected to the network
  - equipped with a terminal emulator application such as Telnet or rlogin
  - equipped with a web browser

- Maintenance telephone, for certain maintenance and testing activities. For more information, see the *Large System: Maintenance* (553-3021-500) guide.
- Maintenance terminals (VDTs and TTYs) with a serial connection to the Succession Call Server, Succession Media Gateways, Succession Signaling Server, or Voice Gateway Media Card.

Input/output terminals can operate either locally or remotely. For local or remote access, maintenance terminal connections can go to the Succession 1000 components through a terminal server, modems, and over the TLAN or ELAN network. Strictly-local access is over a serial cable connected directly to the component in question.

The minimum requirement for a maintenance terminal is a VT100 compatible device. Under some conditions, a null modem adaptor is required to interface the maintenance terminal to the system.

Dial-up modems are used on ports that can establish a PPP link. A single device connection is robust and simple, but a central connection point promotes ease of use. The following methods promote centralized access to a varying degree:

- Modem on the Succession Call Server or Succession Signaling Server with PPP link for Telnet applications and web access for normal operations (not emergency maintenance). This enables access to the ELAN.
- Modem router on the ELAN
- Terminal server
- Secure dial-up Remote Access Server (RAS)
- VPN access to the enterprise network over the Internet

A PC or workstation with a web browser is required for Element Manager.

Modem requirements

A 9600 baud auto-answer modem is the recommended to access the system. A 1200 baud modem is the minimum requirement. You can use the modem to perform service changes, maintenance functions, and diagnostic functions from a remote location. You can perform additional maintenance functions through remote access on the Succession 1000 system. For additional information, refer to *Software Input/Output: Maintenance* (553-3001-511).

Administration tools

Element Manager enables access to the system configuration through a web user interface. Element Manager is accessed directly using a web browser or by using the OTM 2.1 navigator (which includes integrated links to each Element Manager server in a given network).

> ***Note:*** For information about OTM requirements and installing OTM for the Succession 1000 system, refer to *Optivity Telephony Manager: System Administration* (553-3001-330), and *Using Optivity Telephony Manager Release 2.1 Telemanagement Applications* (553-3001-331).

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:

- three 25-pair cables from each Succession Media Gateway
- four 25-pair cables from each Succession Media Gateway Expansion
- four conductors for the AUX cable from the Succession Media Gateway
- one 25-pair cable from each QUA6 PFTU
- wiring from telephone sets and trunks

The BIX cross-connect system is recommended for the Succession 1000 system. However, use of this system is not mandatory. Use other cross-connect systems, if required. For example, use the Krone Test Jack Frame in the UK and the Reichle Masari cross-connect terminal in 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. Refer to *Succession 1000 System: Installation and Configuration* (553-3031-210) for additional information about the BIX, Krone Test Jack Frame, and Reichle Masari cross-connect terminals.

Planning regulatory information

Contents

This section contains information on the following topics:

Introduction

This chapter includes regulatory information for American, Canadian, and International installations of the Succession 1000 system.

System approval

All global markets approve the Succession 1000 system. A regulatory label on the back of system equipment contains national and international regulatory information.

The Succession 1000 system meets Class A EMC requirements for all countries.

CAUTION
In a domestic environment, the Succession 1000 system can cause radio interference. In this case, the user could be required to take adequate measures.

Notice for United States installations

The Succession 1000 system follows Part 68 of the FCC rules. A label containing the FCC registration number and Ringer Equivalence Number (REN) for the Succession 1000 equipment is on the back of the Succession Media Gateway and Succession Media Gateway Expansion. Provide this information to your telephone company, if it requests the FCC registration number and REN information.

Importance of Ringer Equivalence Number

The FCC regulation label includes the Ringer Equivalence Number (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 Succession 1000 equipment causes harm to the telephone network, the telephone company can temporarily discontinue your service. The telephone company can ask you to disconnect the equipment from the network until the problem is corrected and you are sure the equipment is working correctly. If possible, the telephone company notifies you before they disconnect the equipment. You are notified of your right to file a complaint with the FCC.

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

If you experience trouble with your Succession 1000 system, contact your authorized distributor or service center in the United States for repair or warranty information.

Hearing aid compatibility

All proprietary telephones used with the Succession 1000 system meet with the requirements of FCC Part 68 Rule 68.316 for hearing aid compatibility.

Notice for Canadian installations

Industry Canada, formerly known as the Canadian Department of Communications, 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. 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 aforementioned conditions does not always prevent service degradation.

An authorized Canadian maintenance facility, identified by the supplier, must make repairs to certified equipment. 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.

CAUTION
System Failure
The Succession 1000 frame ground of each unit must be tied to a reliable building ground reference.

CAUTION
Damage to Equipment
Contact your local electrical inspection authority or electrician to make electrical ground connections.

Canadian and United States network connections

Provide the information in Table 30 to the local telephone company when ordering standard network interface jacks for the Succession 1000 system.

Note: Table 30 includes columns for system port identification, Facility Interface Code (FIC), Service Order Code (SOC), USOC jack identification, and related Nortel Networks equipment part numbers.

Table 30
Network connection specifications (Part 1 of 2)

Ports MTS/WATS	Facility Interface Code	Service Order Code	REN	Network Jacks	Manufacturer network interface port designation
2-Wire, LSA, L-S (2-Wire, Local Switched Access, Loop-Start)	02LS2	9.0F	1.1B	RJ21X CA21X*	NT8D14
2-Wire, LSA, G-S (2-Wire, Local Switched Access, Ground-Start)	02GS2	9.0F	1.1B	RJ21X CA21X*	NT8D14
2-Wire, LSA, R-B (2-Wire, Local Switched Access, Reverse-Battery)	02RV2-T	9.0F	0.0B	RJ21X CA21X*	NT8D14
1.544 Mbit OSI, SF	04DV9-B	6.0P	N/A	RJ48 CA48*	NTAK09
1.544 Mbit OSI, SF	04Dv9-C	6.0P	N/A	RJ48 CA48*	NTAK09

Table 30
Network connection specifications (Part 2 of 2)

Ports MTS/WATS	Facility Interface Code	Service Order Code	REN	Network Jacks	Manufacturer network interface port designation
Analog PL facilities					
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, 2-wire type 1 E&M)	TL31M	9.0F	N/A	RJ2GX CA2GX*	NT8D15
E&M 4-Wire DRTT (TIE line, lossless, dial repeating, 2-wire type 2 E&M)	TL32M	9.0F	N/A	RJ2HX CA2HX*	NT8D15
Note: * RJ with CA for Canada					

FCC compliance: registered equipment for Direct Inward Dial calls

Part 68 of the FCC's rules states that equipment registered for Direct Inward Dial (DID) calls must provide correct answer supervision. 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.

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 for help.

European compliance information

The Succession 1000 system meets the following European technical regulations: CTR 3, CTR 4, CTR 6, CTR 10, CTR 12, CTR 13, CTR 22, and the EN 301 797. Additionally, the Succession 1000 system meets the following European safety specifications: EN 60825, and EN 60950. Analog interfaces are approved based on national specifications. Digital interfaces are approved based on European specifications.

Electromagnetic Compatibility

Table 31 shows the Electromagentic Compatibility (EMC) specifications for the Succession 1000 system.

Table 31
Succession 1000 EMC specifications

Emission:	EN 55022 EN 300329 ETS 300446
Immunity:	EN 55024

Radio and TV interference

Information for the United States of America

The Succession 1000 system follows Part 15 of the FCC rules in the United States of America. Operation is subject to the following two conditions:

1 The Succession 1000 system must not cause harmful interference.

2 The Succession 1000 system must accept any interference received, including interference that can cause undesirable operation.

If the Succession 1000 system causes interference to radio or television reception, try to correct the interference using the following measures. You can determine the presence of interference by placing a telephone call while monitoring. To do this, move the receiving TV or radio antenna where 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. Also, 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

Information for Canada

The Succession 1000 system Class A digital apparatus complies with Canadian ICES-003.

Planning reliability strategies

Contents

This section contains information on the following topics:

Introduction

Reliability in the Succession 1000 system is based on:

1. The reliability/mean time between failures (MTBF) of components
2. Data Network robustness
3. End-point survivability

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

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

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

- Succession Call Server with automatic database distribution by the way of an Alternate Succession Call Server
- Internet Telephone software running on a VGMC in a load-sharing configuration
- Succession Signaling Server software, including H.323 Gateway and Internet Telephone software
- H.323 Gatekeeper

Response to different points of failure

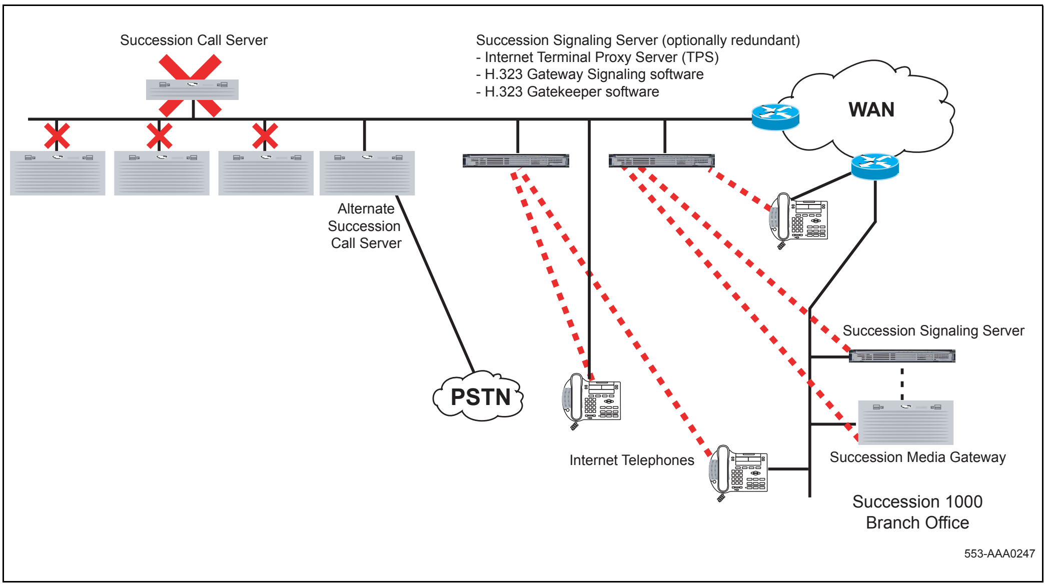
The following topics describe possible failure points and suggested remedies.

- Succession Call Server failure (see page 131)
- Network failure (see page 133)
- Succession Signaling Server failure (see page 134)
- Gatekeeper failure (see page 135)
- Gatekeeper failure – fail-safe (see page 137)

Succession Call Server failure

Figure 23 shows a network-wide view of Succession Call Server failure.

Figure 23
Succession Call Server failure



Alternate Succession Call Server

This situation applies when the Succession 1000 equipment is co-located and not widely distributed.

All of the Succession Media Gateways are equipped with a full set of call processing software components and maintain a configuration database that is periodically synchronized with the primary Succession Call Server.

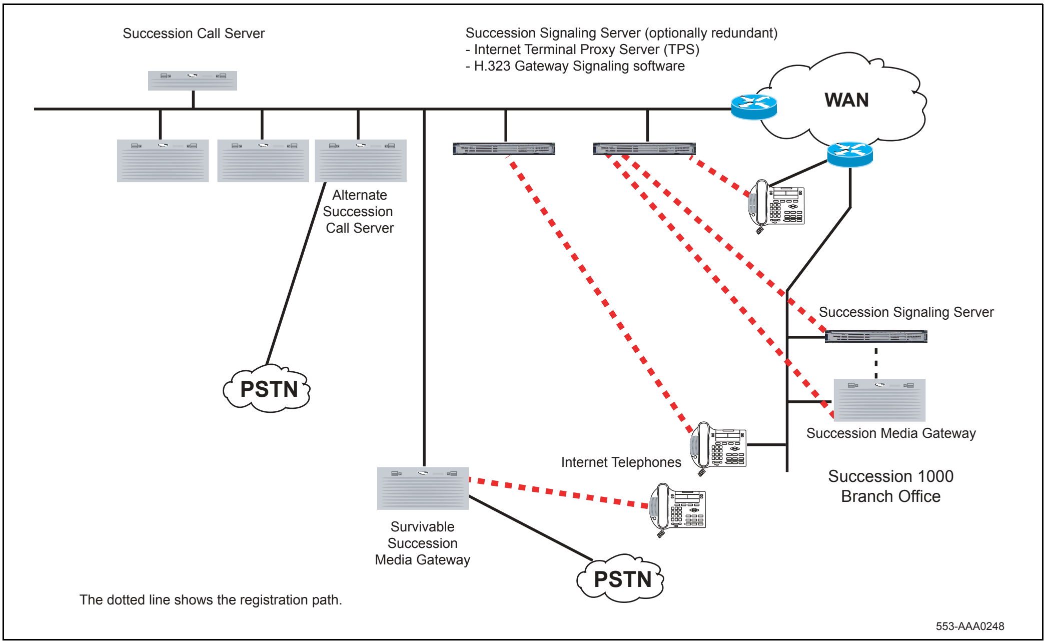
When planning reliability strategies, one Succession Media Gateway should be provisioned as an Alternate Succession Call Server within the IP Telephony node. To support an Alternate Succession Call Server, the installer must configure the Alternate Succession Call Server IP address in Element Manager.

If the Succession Call Server fails, as shown in Figure 23 on page 131, the Succession Media Gateway assigned as an Alternate Succession Call Server assumes the role of the Succession Call Server. The Succession Signaling Servers register to Alternate Succession Call Server and system operation resumes. Operation resumes with single Succession Media Gateway cards, such as analog and PRI cards.

Network failure

Figure 24 illustrates a network failure with Survivable Succession Media Gateways.

Figure 24
Network failure with Survivable Succession Media Gateways



Campus-distributed Succession Media Gateway in Survival Mode

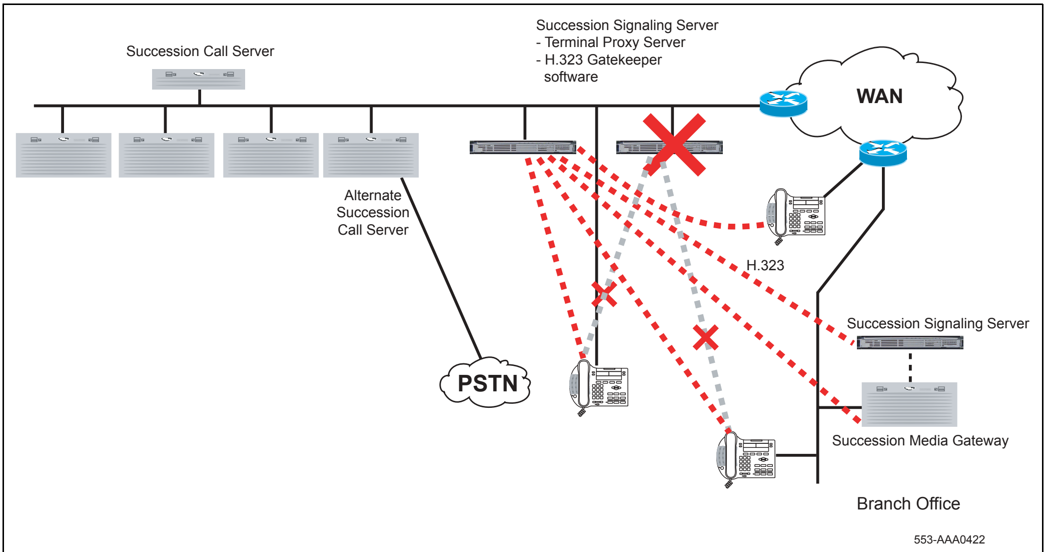
Succession Media Gateways can be configured as survivable when distributed throughout a campus environment. Therefore, basic telephony services can be provided in the event of a network outage. When planning for survivable Succession Media Gateways, consider the location of critical telephones and trunks.

If the network to the IP campus fails, the remote Succession Media Gateway shifts to the Survival Mode. The Internet Telephones register to the TPS on the VGMCs within the remote Succession Media Gateway. The remote Internet Telephones can now access the Main Office over the PSTN trunks.

Succession Signaling Server failure

Figure 25 illustrates the failure of a Succession Signaling Server.

Figure 25
Succession Signaling Server failure



Succession Signaling Server redundancy

Succession Signaling Server redundancy provides a load-sharing basis for the Internet Telephone Terminal Proxy Server (TPS) and an alternate route for the H.323 Gateway software.

When planning Succession Signaling Server survivability strategies, a second or redundant Succession Signaling Server should be installed. As shown in Figure 25 on page 134, two Succession Signaling Servers can load-share when the Succession Media Gateways contain multiple VGMCs. Also, one Succession Signaling Server is a lead Succession Signaling Server that acts as the primary, master TPS. The other Succession Signaling Server is a follower Succession Signaling Server that acts as a secondary, redundant TPS, Virtual Trunk, and Gatekeeper.

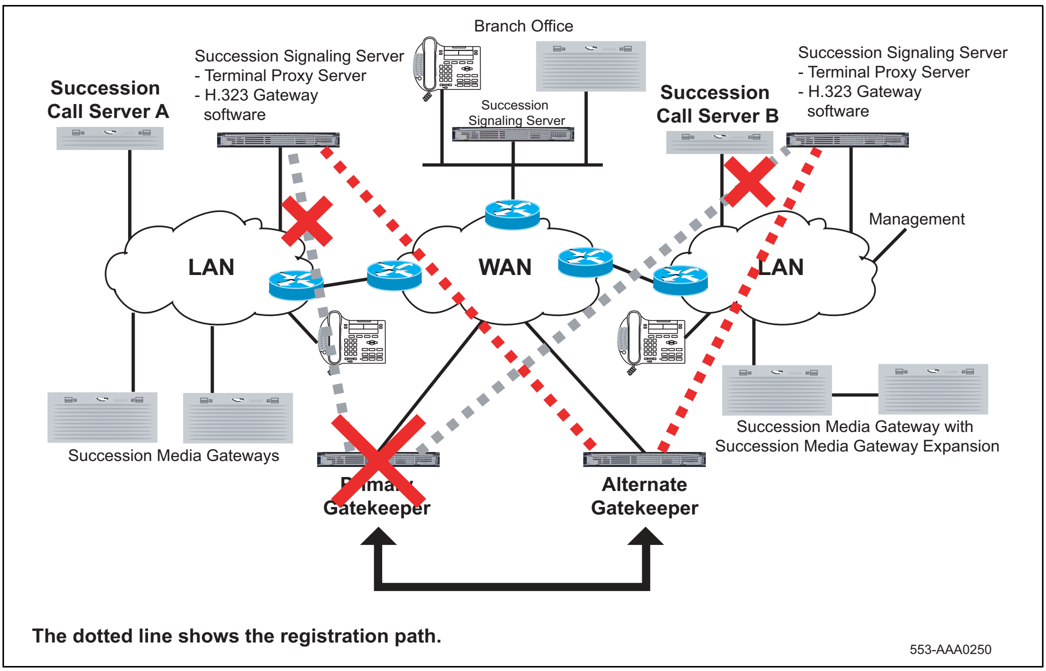
If the lead Succession Signaling Server fails, an election process takes place and the follower Succession Signaling Server becomes the master TPS. The Internet Telephones reregister to the follower Succession Signaling Server and the system operation resumes. If the follower Succession Signaling Server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

> ***Note:*** The same functionality is available without a redundant Succession Signaling Server. Voice Gateway Media Cards in other Succession Media Gateways can assume a TPS role and become a source for Internet Telephone registration.

Gatekeeper failure

Figure 26 illustrates a Gatekeeper failure.

Figure 26
Gatekeeper failure



Gatekeeper redundancy

Figure 26 depicts a distributed environment where the TPS and H.323 Gatekeeper software resides with Succession Call Server A and Succession Call Server B on there own Succession Signaling Server.

> ***Note:*** The H.323 Gatekeeper TSP and the H.323 Gateway software can both reside on a single Succession Signaling Server. Furthermore, Primary software, the TPS, and the H.323 Gateway can all reside on Succession Call Server A, while the second instance of H.323 Gatekeeper software can reside on a separate Succession Signaling Server with the TPS.

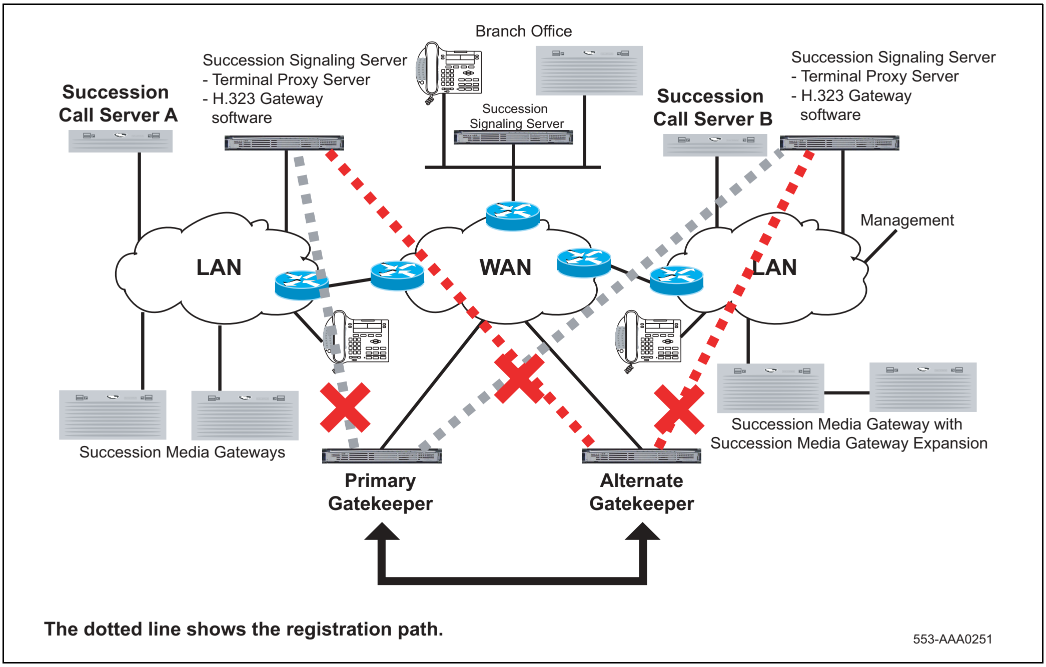
Succession 1000 networks are equipped with at least one H.323 Gatekeeper to provide management of the network numbering plan for private and public numbers. An optional redundant Gatekeeper can be installed in the network. This alternate Gatekeeper automatically synchronizes its database with the primary Gatekeeper periodically.

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

Gatekeeper failure – fail-safe

Figure 27 illustrates Gatekeeper fail-safe.

Figure 27
Gatekeeper failure - fail-safe



Gatekeeper fail-safe

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

The H.323 Gatekeeper can be configured as Primary, Alternate, or Fail-safe. If both Gatekeepers fail or a network outage to a Gatekeeper occurs, the H.323 Gateways route calls using cached data until communication to the Gatekeeper resumes.

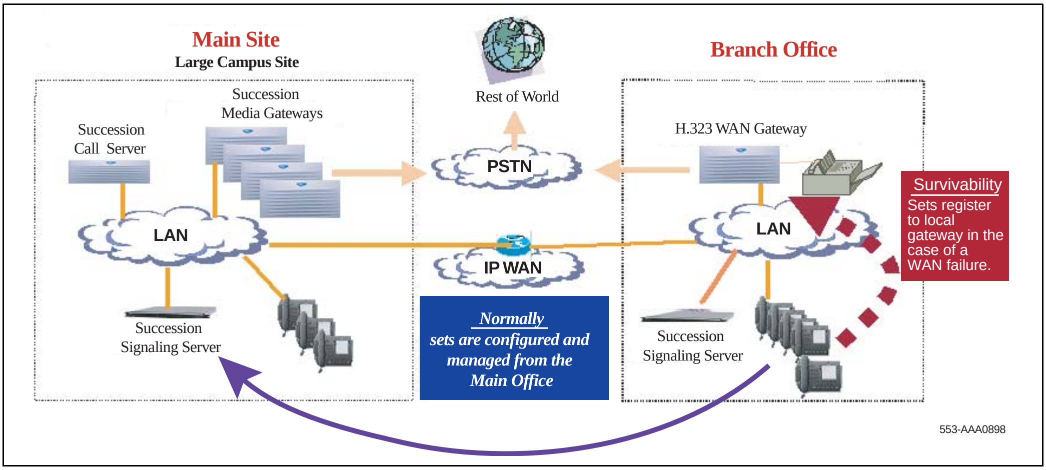
Branch Office survivability

Refer to *Branch Office* (553-3001-214) for details on Branch Office survivability.

Survival mode operation (Local Mode)

If the WAN connection to the Main Office fails, the local Succession Media Gateway can support call handling for the Branch Office Internet Telephones. The Succession System Controller (SSC) processes calls on a per-telephone basis under Local Mode (or under Test Local Mode) when system connectivity is down. In Local Mode, Branch Office Users have full access to local analog or digital trunks. This is known as survivability.

Figure 28
Branch Office



When WAN connectivity is lost, each Internet Telephone loses its registration with the Main Office Terminal Proxy Server (TPS). The Internet Telephone reboots and registers to a TPS on the Voice Gateway Media Card in the Succession Media Gateway.

When locally registered, the Internet Telephones display “Local Mode”. With proper ESN configuration, Branch Office Internet Telephones also access Internet Telephones at the Main Office through the local PSTN.

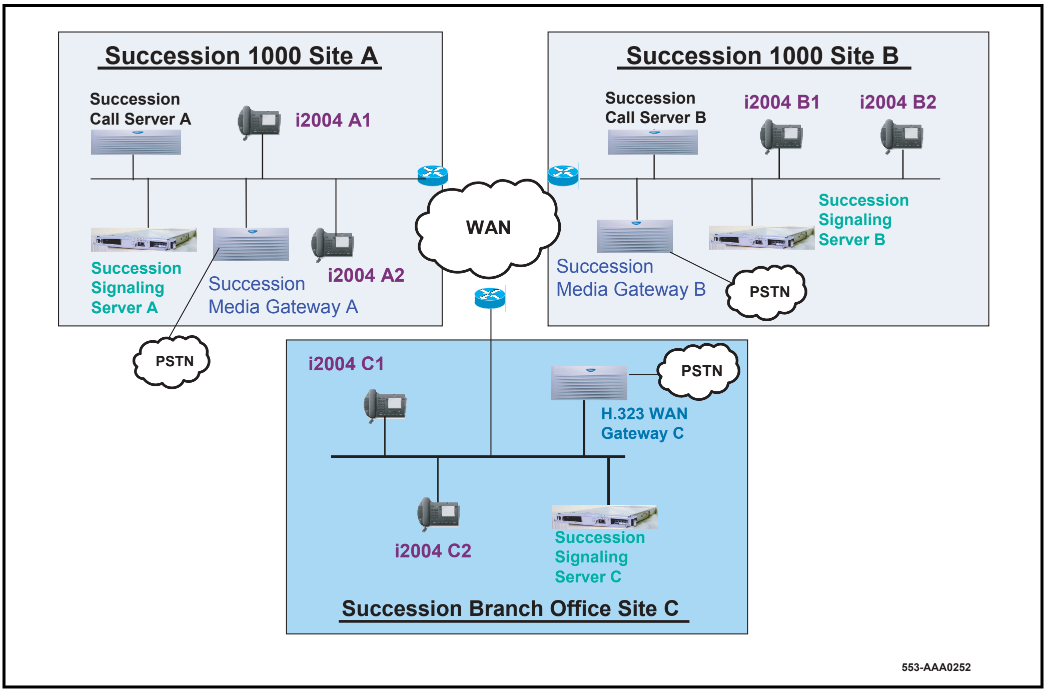
Succession 1000 resiliency scenarios

This section describes the following resiliency scenario categories:

- Succession Call Server failure scenarios (see page 140)
- Succession Signaling Server failure scenarios (see page 143)
- Gatekeeper failure scenarios (see page 147)
- Branch Office scenarios (see page 151)

Refer to Figure 29 when reviewing these scenarios.

Figure 29
Succession 1000



Succession Call Server failure scenarios

Resiliency Scenario 1

i2004 A1 and A2 are talking over the LAN and Succession Call Server A fails.

What happens to the call in progress?

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

Describe what happens:

Succession Media Gateway A reboots and if it is configured as an Alternate Succession Call Server, it begins taking over all call processing. The Succession Signaling Server reregisters to the Alternate Succession Call Server so service can be restored for all Internet Telephones.

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

1.5 minutes for the Succession Media Gateway reboot plus switchover timer. (Default for switchover timer is 2 minutes).

Succession Call Server failure scenarios (Continued)

Resiliency Scenario 2

i2004 A1 and i2004 B2 are talking and Succession Call Server A fails.

What happens to the call in progress?

Same as Scenario 1.

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

Describe what happens:

Same as Scenario 1.

Succession Media Gateway A reboots and if it is configured as an Alternate Succession Call Server, it begins taking over all call processing. The Succession Signaling Server reregisters to the Alternate Succession Call Server so service can be restored for all Internet Telephones.

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

Same as Scenario 1.

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

Succession Call Server failure scenarios (Continued)

Resiliency Scenario 3

i2004 A1 is talking to someone locally or off net over a PSTN trunk in Succession Media Gateway A, and Succession Call Server A fails.

What happens to the call in progress?

Same as Scenario 1.

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

Describe what happens:

Same as Scenario 1.

Succession Media Gateway A reboots and if it is configured as an Alternate Succession Call Server, it begins taking over all call processing. The Succession Signaling Server reregisters to the Alternate Succession Call Server so service can be restored for all Internet Telephones.

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

Same as Scenario 1.

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

Succession Signaling Server failure scenarios

Resiliency Scenario 4

i2004 A1 and A2 are talking over the LAN and Succession Signaling Server A fails. (Assumes no redundant Succession Signaling Server but fail-over TPS to Voice Gateway Media Card (VGMC).)

What happens to the call in progress?

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

Describe what happens:

i2004 Internet Telephones reboot and reregister with the VGMC card.

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

0.5 to 1 minute after the call is dropped. (Assuming Succession Media Gateway A has a VGMC configured so that A1 and A2 can reregister to the VGMC.)

Succession Signaling Server failure scenarios (Continued)

Resiliency Scenario 5

i2004 A1 and i2004 B2 are talking and Succession Signaling Server A fails. (Assumes no redundant Succession Signaling Server but fail-over TPS to VGMC.)

What happens to the call in progress?

Same as Scenario 4.

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

Describe what happens:

Same as Scenario 4.

i2004~~s~~ Internet Telephones reboot and reregister with the VGMC.

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

0.5 to 1 minute after the call is dropped. (Assuming Succession Media Gateway A has a VGMC configured so that A1 and A2 can reregister to the VGMC.)

The call cannot re-initiate exactly as in this scenario, there is no longer a means of setting up a Virtual Trunk session, therefore, the call will be routed out over an alternative route (for example, PRI channel).

Succession Signaling Server failure scenarios (Continued)

Resiliency Scenario 6

i2004 A1 and i2004 B2 are talking and Succession Signaling Server A fails. A redundant Succession Signaling Server is configured on Site A.

What happens to the call in progress?

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

Describe what happens:

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

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

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

Succession Signaling Server failure scenarios (Continued)

Resiliency Scenario 7

i2004 A1 and i2004 A2 are talking and Succession Signaling Server A fails. A redundant Succession Signaling Server is configured on Site A.

What happens to the call in progress?

Same as Scenario 6.

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

Describe what happens:

Same as Scenario 6.

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

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

Same as Scenario 6.

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

Gatekeeper failure scenarios

Resiliency Scenario 8

i2004 A1 and i2004 B2 are talking and the Primary Gatekeeper fails. An Alternate Gatekeeper is configured on Site B. Assume the Primary Gatekeeper is a standalone box (without a TPS).

What happens to the call in progress?

The calls in progress are unaffected.

Describe what happens:

The Alternate Gatekeeper takes over as Active Gatekeeper after the 30 second polling timer expires.

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

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

New calls will establish after:

- the 30 second polling timer expires
- the Alternate Gatekeeper switches over to the Active Gatekeeper
- the Time to Live timer expires

Gatekeeper failure scenarios (Continued)

Resiliency Scenario 9

i2004 A1 and i2004 B2 are talking and the Primary Gatekeeper (Succession Signaling Server) fails. Assume the Primary Gatekeeper is co-resident with the Succession Signaling Server TPS on Site A. An Alternate Gatekeeper is configured on Site B. Assume the Alternate Gatekeeper is co-resident with Succession Signaling Server TPS on Site B. A redundant Succession Signaling Server is configured on Site A.

What happens to the call in progress?

Similar to Scenario 6.

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

Describe what happens:

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

Gatekeeper failure scenarios (Continued)

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

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

For the other 50% of the telephones already registered to the redundant Succession Signaling Server, new calls will establish after:

- the 30 second polling timer expires
- the Alternate Gatekeeper switches over to the Active Gatekeeper
- the Time to Live timer expires

Gatekeeper failure scenarios (Continued)

Resiliency Scenario 10

i2004 A1 and i2004 B2 are talking and both the Primary and Alternate Gatekeepers fail (both are standalone Gatekeepers).

What happens to the call in progress?

Same as scenario 8.

The calls in progress remain unaffected.

Describe what happens:

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

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

Both Primary and Alternate Gatekeeper timers expire. New calls will establish after:

- the 30 second polling timer expires
- the Alternate Gatekeeper switches over to the Active Gatekeeper
- the Time to Live timer expires

Branch Office scenarios

Resiliency Scenario 11

i2004 C1 and C2 are talking over the LAN and Succession Call Server A fails.

What happens to the call in progress?

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

Describe what happens:

Succession Media Gateway A reboots at the Main Office site and acts as an Alternate Succession Call Server at Site A. The Branch Office telephones on Succession Signaling Server A registers with the Alternate Succession Call Server.

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 12

i2004 C1 and A2 are talking and Succession Call Server A fails.

What happens to the call in progress?

Same as Scenario 11.

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

Describe what happens:

Same as Scenario 11.

Succession Media Gateway A reboots at the Main Office site and acts as an Alternate Succession Call Server at Site A. The Branch Office telephones on Succession Signaling Server A register with the Alternate Succession Call Server.

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

Same as Scenario 11.

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

Branch Office scenarios (Continued)

Resiliency Scenario 13

i2004 C1 and C2 are talking over the LAN and Succession Signaling Server A fails.

Will the Branch Office telephones reregister to the redundant Succession Signaling Server at the Main Office or Voice Gateway Media Card (VGMC) at the Main Office (if there is no second Succession Signaling Server)?

What happens to the call in progress?

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

Describe what happens:

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

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 14

i2004 C1 and A2 are talking over LAN and Succession Signaling Server A fails.

Will the Branch Office telephones reregister to redundant Succession Signaling Server at the Main Office or Voice Gateway Media Card at the Main Office (if there is no redundant Succession Signaling Server)?

What happens to the call in progress?

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

Describe what happens:

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

Branch Office scenarios (Continued)

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

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

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 15

i2004 C1 and C2 at the Branch Office are talking and the WAN data network connection to the Main Office goes down.

What happens to the call in progress?

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

Describe what happens:

Telephones C1 and C2 reboot and then reregister with the Succession Signaling Server at the Branch Office.

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 16

i2004 C1 and A2 are talking and the WAN data network connection to the Main Office goes down.

What happens to the call in progress?

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

Describe what happens:

A2 stays registered with Succession Signaling Server A. C1 reboots and registers with Succession Signaling Server at the Branch Office.

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 17

i2004 C1 is talking to someone off net over a PSTN trunk in H.323 WAN Gateway C (Branch Office) and Succession Call Server A fails.

What happens to the call in progress?

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

Describe what happens:

Succession Media Gateway A reboots at the Main Office site and acts as an Alternate Succession Call Server at Site A. The Branch Office telephones on Succession Signaling Server A register with the Alternate Succession Call Server.

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 18

i2004 C1 is talking to i2004 C2 and the Branch Office Succession Signaling Server fails?

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

No impact on existing or future Branch Office IP to IP calls in progress.

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

Not applicable.

Branch Office scenarios (Continued)

Resiliency Scenario 19

i2004 C1 is talking to someone off net over a PSTN trunk in H.323 WAN Gateway C and Succession Signaling Server C (Branch Office) fails. (The behavior is the same as i2004 A1 talking to someone off net over a PSTN trunk in Succession Media Gateway B and Succession Signaling Server B fails).

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

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

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

If there is no redundant Succession Signaling Server in the Branch Office, calls of this type cannot be initiated until the Succession Signaling Server is re-established. The call would, in this instance, be routed out over an alternative PSTN route.

Branch Office scenarios (Continued)

Resiliency Scenario 20

A digital telephone in the Branch Office is talking to someone off net over a PSTN trunk in H.323 WAN Gateway C and Succession Signaling Server C (Branch Office) fails.

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

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

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

Not applicable.

Branch Office scenarios (Continued)

Resiliency Scenario 21

A digital telephone in the Main Office is talking to someone off net over a PSTN trunk in H.323 WAN Gateway C and Succession Signaling Server A (Main Office) fails. A redundant Succession Signaling Server is installed at Site A. (This is the same as a digital telephone in Succession Media Gateway A is talking to someone off net over a PSTN trunk in Succession Media Gateway B and Succession Signaling Server A fails).

What happens to the call in progress?

No impact on the call in progress.

Describe what happens:

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

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 22

A digital telephone in the Main Office Site A is talking to someone off net over a PSTN trunk in H.323 WAN Gateway C and Succession Signaling Server A (Main Office) fails. No redundant Succession Signaling Server is installed at Site A. PSTN is configured as an alternate route. (This is the same as a digital telephone in Succession Media Gateway A talking to someone off net over a PSTN trunk in Succession Media Gateway B and Succession Signaling Server A fails).

What happens to the call in progress?

No impact to the call in progress.

Describe what happens:

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

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 23

A digital telephone in the Main Office Site A is talking to a digital telephone in H.323 WAN Gateway C and Succession Signaling Server A (Main Office) fails. No redundant Succession Signaling Server is installed at Site A. PSTN is configured as an alternate route. (This is the same as digital telephone in Succession Media Gateway A is talking to digital telephone in Succession Media Gateway B and Succession Signaling Server A fails).

What happens to the call in progress?

No impact to the call in progress.

Describe what happens:

All idle Virtual Trunks become idle unregistered. Virtual Trunks that are busy on established calls also become unregistered, but they remain busy until the calls are released. There is no impact on the media path between the DSP connected to digital telephone in Main Office Site A and that connected to digital telephone in H.323 WAN Gateway C.

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

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

Branch Office scenarios (Continued)

Resiliency Scenario 24

A digital telephone in the Main Office Site A is talking to Internet Telephone C1 in H.323 WAN Gateway C and Succession Signaling Server A (Main Office) fails. No redundant Succession Signaling Server installed at Site A. Voice Gateway Media Cards installed in Site A.

What happens to the call in progress?

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

Describe what happens:

i2004 Internet Telephones reboot and reregister with Voice Gateway Media Cards at the Main Office site by way of the Branch Office TPS.

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

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

System capacity

Contents

This section contains information on the following topics:

Introduction

This chapter describes the capacity of the Succession 1000 system. System or network capacity can be determined by the following:

- physical capacity
- software capacity limit
- application limit (practical limit)

This section uses the most relevant limit to outline the capacity of a given item. It is to be used as a quick overview, for planning purposes. For detailed calculations, please refer to the following chapters:

- "Mixed Call Center and Main Office example" on page 205
- "Branch Office example" on page 215

Note: Real-time capacity must also be considered for the specific application, and can also constrain any applications in reaching resource limits.

Table 32
System practical application limits (Part 1 of 6)

Item	Practical limit	Comment
System Virtual TNs	1248†	The Virtual TNs are used for: • Internet Telephone • Virtual Trunks • Phantom TNs
Internet Telephones	1000 for each Succession Call Server	Each Internet Telephone requires a Virtual TN. Internet Telephones in the Branch Office must be included in this practical limit. Larger line size can be achieved by networking multiple Succession Call Servers. ***Note:*** The limit cannot be extended by engineering.
Digital Telephones	448	The practical limit of 448 telephones is based on physical capacity of the main system, assuming a 10% trunking ratio, and CallPilot installed in the system. The addition of a Branch Office extends the digital telephone capacity. The practical limit assumes the system has 32 card slots available in Succession Media Gateways and Succession Media Gateway Expansions. Given two slots for CallPilot, there is a maximum of 30. Assuming all telephones are digital telephones (16 ports/card). A possible maximum configuration is 28 line cards and two PRI cards. This gives 448 telephones and 48 trunks, which equates to approximately 10% trunking. The theoretical limit is 512 digital ports based on physical slot capacity of Succession Media Gateways. This assumes only digital telephone lines are installed in the system.

Table 32
System practical application limits (Part 2 of 6)

Item	Practical limit	Comment
Analog Devices (telephones/ trunks)	448	The practical limit is 448 telephones based on physical capacity of the main system, and assuming a 10% trunking ratio, and CallPilot installed in the system. This assumes the system has 32 card slots available in Succession Media Gateways and Succession Media Gateway Expansions. Given two slots for Callpilot, there is a maximum of 30. Assuming all devices are analog (16 ports/card). A possible maximum configuration is 28 line cards and two PRI cards. This gives 448 sets and 48 trunks (about 10% trunking).
E1 PRI trunks	480	Four PRI cards for each Succession Media Gateway = 16 for each system.The addition of a Branch Office extends the E1 PRI trunk capacity. At 30 ports maximum (E1), there are 480 digital trunks.
T1 PRI trunks	384	Four PRI cards for each Succession Media Gateway = 16 for each system. The addition of a Branch Office extends the T1 PRI trunk capacity. At 24 ports maximum (T1), there are 384 digital trunks.

Table 32
System practical application limits (Part 3 of 6)

Item	Practical limit	Comment
Virtual Trunks on a Succession Signaling Server	382†	The maximum number of Virtual Trunks for each Succession Call Server is 1248, which equals the maximum number of Virtual TNs in the system. A Virtual TN is either an IP telephone or a Virtual Trunk. Therefore, the limit of 1248 Virtual TNs is the sum of both Internet Telephones and Virtual Trunks. Up to 382 Virtual Trunks are supported on a Succession Signaling Server for each IP Telephony node. Additional Virtual Trunks can be supported with the addition of one Succession Signaling Server for a total of 764 Virtual Trunks. ***Note:*** Each Succession Signaling Server must have it's own Trunk Route.
ACD Agent telephones	300	To configure digital telephones or Internet Telephones as ACD agent telephones, enough trunks are needed to carry incoming traffic. If the maximum number of trunks is 384, the realistic number of agent telephones is 300, leaving room for queuing.
Voice Gateway Media Cards (VGMCs)	15	A VGMC has 32 ports. Its transcoding traffic must come from trunk or line cards. The addition of a Branch Office extends the VGMC capacity. The highest density traffic source is an E1 PRI of 30 ports. To balance traffic from TDM channels to transcoding ports within a maximum of 30 card limitation, one practical configuration is 15 E1 PRI of 450 TDM traffic sources to 15 VGMCs (480 ports), which roughly balances out TDM channels and transcoding ports without network internal blocking.

Table 32
System practical application limits (Part 4 of 6)

Item	Practical limit	Comment
CallPilot (201i IPE)	2	A CallPilot provides up to 40 IVR ports. At P.01, it can handle 1044 CCS. A 1000-telephone system typically generates 1000 CCS voice message traffic. Therefore, one CallPilot is generally enough to serve most applications. However, should a heavy IVR application generate more than typical message traffic, at most two CallPilot systems should be enough. These two systems have 80 combined ports, but function as two 40-port systems.
DCH ports for each trunk route	1	One DCH port must be configured for every 200 Virtual Trunks, or one for each trunk route. If more than one trunk route is configured in the system, additional DCH ports must be configured. No physical SDI/DCH card is required for Virtual Trunks.
Succession Call Server rated capacity	35,000 EBCs per hour	A Basic Call is a 2500 analog telephone to another 2500 telephone call within the same switch. All other types of calls are feature calls, which take more processing time than a Basic Call. A feature call can be multiplied by a real-time factor to become Equivalent Basic Calls (EBC). The total EBCs of a system should not be greater than the Server rated capacity.
Succession Call Server IP call capacity	15,000 calls per hour	The number of unfeatured IP calls (Internet Telephone to Internet Telephone) on the same switch. There is an Internet Telephone real-time factor which can be used to convert IP calls into EBCs for determining the Call Server rated capacity.
Succession Signaling Server IP call capacity	60,000 calls per hour	All Terminal Proxy Server, H.323 proxy, and Gatekeeper functions are co-located on the Succession Signaling Server.

Table 32
System practical application limits (Part 5 of 6)

Item	Practical limit	Comment
Other applications cards		No practical limits.
DTI trunks	384	With 24 ports for each card and 16 cards maximum, the potential limit of Digital Trunk Interface (DTI) is 384 trunks. The addition of a Branch Office extends the DTI trunk capacity.
Analog trunks	192	With 8 ports for each trunk card, 4 cards can generate 32 ports of traffic to the VGMC's 32 ports (4 to 1 card ratio). The addition of a Branch Office extends the Analog trunk capacity. The system maximum of 24 analog trunk cards against 6 VGMCs, resulting in a maximum of 192 analog trunks in the system.
ACD agent sets for each E1 trunk	375	Using 480 trunks as the system limit for E1 PRI trunks, and keeping the trunk to agent ratio (1.28) the same as that for T1, the result is 375 ACD agent sets, leaving a few extra trunks for queuing calls.
IP wireless portable telephones	408	An ITG-W for wireless portable has 24 ports. Assuming the available 30 card slots are split 17 for ITG-W cards and 13 for VGMCs, the resulting configuration has 408 (17 x 24) portables. This number varies depending on how many VGMCs are required.
Companion DECT wireless handsets	480	Among 30 card slots, 15 are used for DECT mobility cards, and 15 are used for VGMCs. The resulting configuration has 480 (32 x 15) wireless handsets. The maximum number varies according to the requirement of VGMCs in the configuration.

Table 32
System practical application limits (Part 6 of 6)

Item	Practical limit	Comment
Remote Office telephones for each Reach Line Card (RLC)	32	For each 32-port Reach Line Card equipped in the Succession 1000 system, up to 32 digital telephones can use the Remote Office 9150 unit to reach a Succession 1000 system or make local calls. ***Note:*** The maximum number of possible Digital, Analog, and Wireless IP portable handsets can vary depending on the type of required trunking. ***Note:*** For Internet Telephones or IP wireless portable handsets, the system set limit can be increased if trunking is through Virtual Trunks instead of PRI/DTI trunks, because Virtual Trunks require no VGMC DSP resources.
Telephones for each Gatekeeper	10,000†	When one Gatekeeper supports multiple Succession Call Server systems.
H.323 endpoints for each Gatekeeper	2,000†	When one Gatekeeper supports multiple Succession Call Server systems or Branch Office systems or both systems. The addition of a Branch Office extends the H.323 endpoints capacity. Each H.323 zone requires one Primary Gatekeeper. An Alternate Gatekeeper can provide redundancy.
Registered Internet Telephone users for 8-port VGMCs	32	The Internet Telephone only registers to these VGMCs when the Succession Signaling Server with the TPS fails.
Registered Internet Telephone users for 32-port VGMCs	128	The Internet Telephone only registers to these VGMCs when the Succession Signaling Server with the TPS fails.
Note: † Denotes an absolute limit.		

Minor application or service cards (tone detector, Digitone receiver, RAN, Music, Paging, and others) are equipped according to engineering requirements. With the maximum number of system devices at 1248, the service card requirement is typically quite small (one to three cards). As long as the calculation is based on engineering model, service cards should not become system bottlenecks.

Major system parameter limitations

In addition to physical equipment impacting system capacity, certain software parameters or applications can constrain the system application limits as follows:

- **Phantom TN**: 224 or 256 minimum, with subset 32 as an increment for DECT or phantom TNs. For example, 224 or 256 phantom TNs configured, 64 TNs for DECT, 96 as the phantom TNs in a call center.
- **Virtual TN**: 256 minimum with subset 32 as an increment for Internet Telephones and Virtual Trunks. For example, 256 TNs configured, 224 for Internet Telephones, 32 for Virtual Trunks.
- **Companion DECT**: 32 wireless sets per DECT mobility card with sufficient antennas - traffic dependent.
- **IP Wireless portable handsets (Symbol)**: 24 wireless handsets per ITG-W card with sufficient Access Points.
- **Virtual Trunks**: A maximum of 382 Virtual Trunks per Succession Signaling Server, all 1248 virtual TNs can be configured for Virtual Trunks if other system resources (Succession Call Server, DSP, and so on) permit.
- **Maximum number of Branch Offices**: No theoretical maximum; the practical, lab-tested maximum is 256. Internet Telephones at Branch Offices count toward the 1000-total of the Main Office virtual TN count.
- **Maximum number of Internet Telephones to register at the same time**: 1000 Internet Telephones. The keymap download time is six to eight minutes.

  Note: The Succession 1000 system only supports ac power and 100BaseT.

Succession Media Gateways

The Succession 1000 system supports up to four Succession Media Gateways and four Succession Media Gateway Expansions.

Serial ports

The maximum number of serial ports used for such things as maintenance, administration, CDR, Traffic, depends on the number of Succession Media Gateways equipped. There are three serial ports on the Succession Call Server and another three serial ports for each equipped Succession Media Gateway.

Branch Offices

There must be a Succession Signaling Server at the Main Office site. One Succession Signaling Server is required for each Branch Office node. Each Branch Office can support up to 400 Internet Telephones. However, this counts against the maximum for the Main Office, which is 1000.

A Branch Office supports one Succession Media Gateway with up to four cards and one Succession Media Gateway Expansion with up to four cards. There are 30 default Virtual Trunks on a Branch Office. The Branch Office can support up to 92 T1 trunks or 120 E1 trunks. The maximum is 256. In practice, the number is much smaller due to the limit at the Main Office which of 1000, including all telephones supported at Branch Offices.

For example, a Main Office with 500 Internet Telephones can support up to 10 Branch Offices with 50 Internet Telephones in each Branch Office.

Networking Succession Call Servers

A Gatekeeper can support up to 10,000 Internet Telephones, which can be associated with up to 2000 H.323 endpoints. A Succession Call Server or a Branch Office are counted as one endpoint.

A Succession Call Server can serve up to 1000 Internet Telephones and 200 Virtual Trunks. The Virtual Trunk number can be higher, if multiple Succession Signaling Servers are equipped. When a larger system is desired, network several Succession Call Servers together through LAN or WAN connection under the control of one Gatekeeper. With MCDN networking software, an IP network can provide network-wide feature transparency.

Since a networked system is comprised of distributed networks, there is no central control in the system for call processing. An Internet Telephone to Internet Telephone call between two Succession Call Servers within a system is treated as a trunk call to each Succession Call Server. The trunk call to each Succession Call Server requires Virtual Trunks. A Virtual Trunk for the originating side Succession Call Server, and a Virtual Trunk for the terminating side Succession Call Server. Double Virtual Trunk resources are used for intrasystem, inter-Succession Call Server calls.

To maximize networking efficiency, Internet Telephones with a known community of interest should be registered on the same Succession Call Server. Avoid networking until a Succession Call Server is fully equipped with maximum number of telephones and trunks. For practical considerations, a fully equipped system with 10,000 Internet Telephones and one Gatekeeper can support 10 Succession Call Servers.

System engineering procedure

Contents

This section contains information on the following topics:

Introduction

In this chapter the following abbreviations are used in the formulas:

- IT = Internet Telephone
- AT = analog (500/2500-type) telephone
- MC = Voice Gateway Media Card (VGMC)

Required application resources

The following application resources are required by the system: OTM management tool, Symposium Call Center Server, CallPilot 201i IPE version, and RAN trunks. Note that applications do not figure into the engineering calculations.

A Symposium Call Center Server (SCCS) is engineered according to SCCS engineering guides. The number of calls receiving SCCS treatment are a known value in Succession 1000 engineering.

CallPilot 201i (IPE version) port requirement is performed during application engineering of a CallPilot. The number of calls receiving CallPilot treatment is a given in Succession 1000 engineering.

RAN trunks installed in Succession Media Gateway/Succession Media Gateway Expansion card slots are typically a part of telephone and trunk traffic. RAN trunks are calculated in port traffic specification. Both RAN trunks and CallPilot ports are not calculated for traffic impact on VGMCs, but are included in real-time impact and card slot calculation.

Calculating traffic

The following procedures can be used to calculate traffic from various sources:

- “Calculating traffic on the Voice Gateway Media Card port” on page 179
- “Calculating traffic on the TLAN” on page 180

Procedure 3
Calculating traffic on the Voice Gateway Media Card port

1. Obtain the telephone traffic from the end-user or use the default value in Table 33 on page 179 from the application model.
2. Multiply telephone traffic by the total number of telephones.
   Total system CCS = Average telephone CCS x Total number of telephones.
3. Determine the Internet Telephone to Internet Telephone traffic.
4. Remove the Internet Telephone to Internet Telephone traffic from VGMC port traffic calculation. MC CCS = Total system CCS – IT CCS.

Table 33
Application model default values

Item	Value
Telephone, including 10% for voice mail (see Note 1)	6 CCS
Telephone in a hotel/motel application	3 CCS
Analog trunk or digital trunk	25 CCS
Telephone average hold time for each call	120 seconds
Trunk hold average time for each call	180 seconds
Telephone/Trunk average hold time for each call	150 seconds
Agent telephone loading to 92%	33 CCS
Note 1: Telephone traffic is 50% incoming calls and 50% outgoing calls, that is 3 CCS in and 3 CCS out. ***Note 2:*** In some Call Center applications, agent calls can be 100% incoming, 100% outgoing for a Call Center equipped with auto-dialers, or a mixed percentage of incoming and outgoing calls.	

End of Procedure

Procedure 4
Calculating traffic on the TLAN

1 Total all Internet Telephone to Internet Telephone traffic, and add Internet Telephone traffic to VGMC port traffic.

Total TLAN CCS = Total IT to IT CCS + IT CCS to MC

2 Divide the total CCS by 36 to convert to Erlangs for TLAN calculation later.

Total system Erlangs = Total TLAN CCS ÷ 36

The following do not generate VGMC traffic:

- Internet Telephone to Internet Telephone calls
- Internal TDM calls within the node
- ISDN PRI and DTI incoming (see note1)
- RAN returned to the caller (see note 2)
- Music returned to the caller (see note 2)
- CallPilot treatment (see note 3)

Note 1: When an ISDN PRI or DTI channel and an Internet Telephone are connected, this generates VGMC traffic.

Note 2: If a call originates on a local trunk routed to RAN, the call does not generate VGMC traffic. If the call originates locally from an Internet Telephone or Virtual Trunk from another node, the call generates VGMC traffic when providing RAN or Music.

Note 3: Only calls from non-Internet Telephones connect to CallPilot without going through a VGMC. If calls are from IP sources, they should use G.711 encoding. Otherwise, double transcoding. For example, once at the Internet Telephones or trunk with G.729A and at CallPilot with proprietary scheme, with different compression ratios renders the quality of voice message unacceptable.

End of Procedure

Determining requirements for ports and cards on the Voice Gateway Media Card

Grade of Service (GOS) is expressed as **P**. The symbol **P** indicates the number of calls out of 100 that can be blocked. For example, P.01 means 1 call out of 100 can be blocked. For more information on GOS, refer to page 203.

For a private network or local network application (non-toll), the P.01 GOS is sufficient. For wider areas extending to traditional toll regions, P.001 GOS is recommended. Table 34 shows the maximum CCS rate on a VGMC and PRI meeting the specified GOS requirement.

Table 34
Traffic capacity at P.01, P.005, and P.001 GOS

Device	P.01	P.005	P.001
VGMC (32-port)	794 CCS	744 CCS	655 CCS
T1 (24 ports)	550 CCS	511 CCS	441 CCS
E1 (30 ports)	732 CCS	685 CCS	600 CCS

For a call center application, non-blocking at the VGMC is recommended. To ensure non-blocking, the number of registered active agents on a VGMC must not exceed the number of ports on the card. Network blocking can happen if more PRI channels try to access the VGMC gateways than there are ports.

> ***Note:*** Using a 24-port or 30-port module to calculate T1/E1 capacity is conservative. Multiple T1s/E1s to one destination are treated as a single group. A single group with a number of T1s/E1s to one destination are more efficient (with higher capacity) than the sum capacity of several groups, with the same number of T1/E1 to the same destination. For example, 48 channels in one route carry more traffic than two routes of 24 channels, at the same GOS.

A T1 of 23B+D, and DTI of 24B are assumed to both handle 24 channels. The impact of D-channel on T1 is lessened when multiple T1s are involved and using nB+D feature (multi-T1s sharing one D channel).

All PRI and VGMC ports are pooled resources in the system. When the circuit groups are large (more than three PRIs or VGMCs), refer to a standard Erlang B table with the specific GOS required to estimate trunk capacity. This provides a more accurate estimation than using the single card method shown in Table 34 on page 181.

Calculating PRI trunk requirements

The following topics are used to calculate trunk requirements for Succession 1000M and Meridian 1 (ESN) and PSTN.

Table 35
Calculating PRI trunk requirements

Calculation	Formula
PRI cards required for VGMC port traffic to private or public network.	PRI cards = MC CCS ÷ CCS for each card where: CCS for each card = 550 CCS for P.01, 441 CCS for P.001
Voice Gateway Media Cards required if the number of VGMCs is the output of the traffic model instead of T1s.	VGMCs = MC CCS ÷ CCS where CCS for each card = 794 CCS for P.01 or 655 CCS for P.001
VGMCs required to support agents. Agent service for a Call Center application is based on the CCS for an agent (typically 30 to 33 CCS) that is specified by the end-user. Non-blocking between VGMCs and agent telephones is recommended. Each VGMC has 32 ports.	VGMCs = Number of agents ÷ 32
Agent traffic to the PRI or other ports of the system (assuming the number of agents is known).	VGMCs = Number of agents x 33 CCS
Agent traffic to the PRI or other ports of the system (assuming the number of agents is unknown).	VGMCs = (PSTN CCS + ESN CCS) ÷ 33 CCS

Calculating Virtual Trunk and LAN/WAN requirements

Virtual Trunks can be used in place of, or in addition to, PRI. Use Table 36 to calculate Virtual Trunk requirements separately.

Table 36
Calculating Virtual Trunk requirements

Calculation	Formula
Virtual Trunk requirements (assuming traffic table is available) Refer to a standard Poisson table to find the number of Virtual Trunks required. Do not exceed the maximum of 200 Virtual Trunks.	Virtual Trunk (VT) CCS = total system IT CCS x% traffic going to or coming from the VT
Virtual Trunk requirements (assuming traffic table is not available) The formula assumes that Virtual Trunks handle traffic at the trunking efficiency of a 30-circuit trunk group. This assumption is also used to calculate E1 PRI cards, where LAN bandwidth = (VT CCS ÷ 36) x 95 kbps and WAN bandwidth = (VT CCS ÷ 36) x 40 kbps.	Virtual Trunk (VT) CCS = VT CCS x (30/732), for P.01 GOS

Note: Refer to Table 40 on page 193 and Table 41 on page 194 for different assumptions of Codec and payload size. The above equations are based on G.711/20 ms, which is conservative and may not apply to every scenario.

Calculating Succession Media Gateway and Succession Media Gateway Expansion requirements

The following procedures and data are used to calculate Gateway chassis requirements.

Card slot capacity

Based on configuration input, use the information in Table 37 on page 185 to allocate cards into the four card slots available in both the Succession Media Gateway and Succession Media Gateway Expansion.

Note 1: A Succession Media Gateway with a digital trunk must have a clock controller installed.

Note 2: Table 37 on page 185 uses the following abbreviations:

MG = Succession Media Gateway
MGE = Succession Media Gateway Expansion

Table 37
Card capacity (Part 1 of 2)

Card type	Card ports	Card slots	Located in	Maximum number in MG	Maximum number in MGE	Comments
VGMC	8 or 32	1	MG/MGE	4	4	Transcoding between TDM and packets. (see Note)
XALC	16	1	MG/MGE	4	4	Used for analog (500/2500-type) telephone, fax, modem
XDLC	16	1	MG/MGE	4	4	PC based attendant console and digital telephones
Analog trunk	8	1	MG/MGE	4	4	Used for analog trunk, RAN, Music, Paging
PRI	24(T1) 30(E1)	1	MG	4	Not supported	Used for Digital trunks for E1 and T1
SDI/DCH	4	1	MG	1	Not supported	Used for Signaling channel for ISL

Table 37
Card capacity (Part 2 of 2)

Card type	Card ports	Card slots	Located in	Maximum number in MG	Maximum number in MGE	Comments
201i CallPilot	40	2	MG/MGE	2	2	Limited by traffic; independent units
Symposium	1000	n/a	n/a	n/a	n/a	100BaseT interface

Note: The recommended number of devices for cards are as follows:

- Register Internet Telephones for each VGMC according to capacity in Table 34 on page 181 and the default CCS for each telephone used. For example, register 124 telephones when P.005 and 6 CCS/telephone (=744/6) are used.
- Register 32 ACD agent Internet Telephones for each VGMC.
- The DSP ports or DS-0 channels in an Succession Media Gateway or Succession Media Gateway Expansion map to the120 time slots of a virtual superloop for transcoding. Therefore, the sum of total ports equipped in an Succession Media Gateway or Succession Media Gateway Expansion must not exceed 120.

Equipment allocation guidelines

Equipment allocation principles are as follows:

- Minimize the number of chassis used.
- Install cards in empty card slots before equipping the next chassis.
- Do not exceed the limits shown in Table 37 on page 185.

Devices and applications not supported on Succession 1000

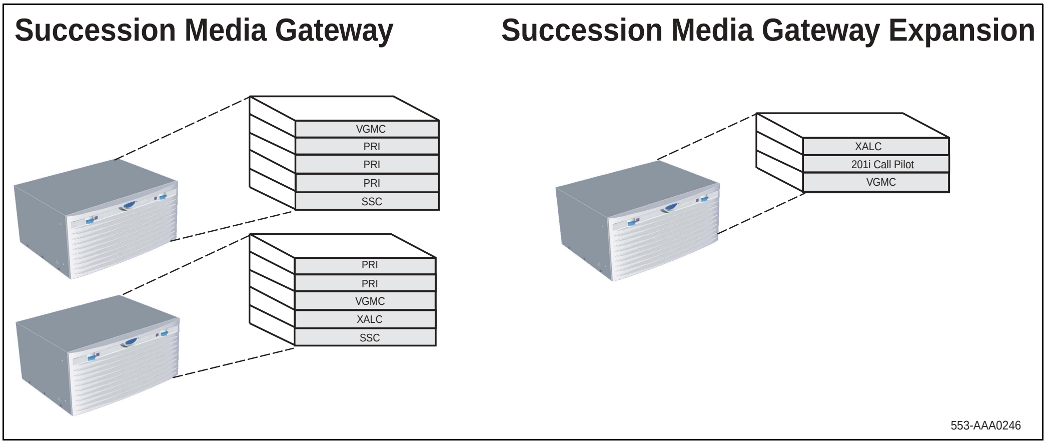
The Succession 1000 system does not support Meridian Mail, a 48-port digital line card, 100BaseF, and Companion wireless.

Example: Succession Media Gateway and Succession Media Gateway Expansion allocation

Figure 30 on page 187 shows a possible configuration for:

- three Voice Gateway Media Cards (VGMC)
- five PRI
- two XALC
- one 201i CallPilot

Figure 30
Example configuration



Card slot assignment

The Succession Media Gateway and Succession Media Gateway Expansion contain physical card slots numbered 1 to 10.

> ***Note:*** Slot 0 is dedicated to the SSC card and is not configurable.

When configuring the Succession 1000 system, the physical card slot numbers must be transposed to “logical” card slot numbers (see Table 38 on page 188). For example, to configure a card physically located in slot 2 of the first Succession Media Gateway, use logical slot 12. To configure a card physically located in slot 2 of the second Succession Media Gateway, use logical slot 22.

Table 38
Succession Media Gateway and Succession Media Gateway Expansion card slot assignments

	Succession Media Gateway / Succession Media Gateway Expansion							
	First		Second		Third		Fourth	
	Physical card slot	Logical card slot	Physical card slot	Logical card slot	Physical card slot	Logical card slot	Physical card slot	Logical card slot
Succession Media Gateway	1	11	1	21	1	31	1	41
	2	12	2	22	2	32	2	42
	3	13	3	23	3	33	3	43
	4	14	4	24	4	34	4	44
	5	*	5	*	5	*	5	*
	6	*	6	*	6	*	6	*
Succession Media Gateway Expansion	7	17	7	27	7	37	7	47
	8	18	8	28	8	38	8	48
	9	19	9	29	9	39	9	49
	10	20	10	30	10	40	10	50

Legend
* Not supported.

Calculating Succession Call Server load

Real-time capacity (load) of a Succession Call Server depends on the number and type of calls that it handles. The following equation can be used to convert CCS to calls:

Calls = CCS for Succession Media Gateway x 100 ÷ Holding time in seconds

The Succession Call Server load can be calculated using total CCS of different call types from earlier calculations using the following formula:

Total EBC = nIT to IT x (1+f1) + nIT to TIE x (1+f0+f2+f5+f6) + nIT to AT x (1+f3) + NIT x (1+f4) + nIT to VT x (1+ f11)

Where:

- Total EBC = Succession Call Server load
  See "Equivalent Basic Call" on page 190.
- nIT to IT = number of Internet Telephone to Internet Telephone calls
- nIT to TIE = number of tie trunk calls served by ACD agents using Internet Telephones
- nIT to AT = number of Internet Telephone to analog (500/2500-type) telephone calls
- nIT to VT = number of Internet Telephone to Virtual Trunk calls
- NIT = non-Internet Telephone calls
- f0 to f11 = See Table 39 on page 191 for the values f0 to f11.

Equivalent Basic Call

The real-time capacity of a switch can be specified in terms of Equivalent Basic Calls (EBC).

> ***Note 1:*** Real-time capacity: The ability of the Succession Call Server to process instructions resulting from calls that meet service criteria.

> ***Note 2:*** Basic Call: A call without features between two 2500 telephones on the same switch. Any other call is a Feature Call.

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

The symbol **fi** is the ratio of a featured call to a basic call excluding the basic call. The **i** in the symbol **fi** can be any of the values 0 to 11, as shown in Table 39 on page 191. The processor capacity is denoted by the EBC at rated capacity. All featured calls in a system are converted into EBC to determine whether the total sum exceeds the designated EBC capacity of a processor. The rated capacity of the Succession Call Server is measured at 35,000 EBC.

The recommended major real-time factors are as shown in Table 39 on page 191.

Table 39
Sample real-time factor

Call Type	Description	Real-Time Factor	Comment
f0	In-bound ACD call	0.13	Added to call center calls
f1	Two-way Internet Telephone to Internet Telephone	1.15	Internet Telephone to Internet Telephone call (no VGMC involved)
f2	One-way Internet Telephone to digital trunk	0.68	Internet Telephone to PRI
f3	One-way Internet Telephone to analog (500/2500-type) telephone or analog trunk	0.48	Internet Telephone to analog (500/2500-type) telephone or trunk
f4	Non-Internet Telephone to digital trunk	0.18	No Internet Telephone involved in the call
f5	In-bound routed to RAN or Music by SCCS	2.06	In-bound call routed to RAN or Music cycle by Symposium
f6	In-bound to IVR then transferred by CallPilot	3.68	Call interacts with IVR then transferred to an agent
f7	In-bound routed to RAN trunk or Music trunk	0.63	Queued incoming call given RAN or Music by RAN trunk
f8	In-bound routed to RAN or Music or IVR by SCCS	5.74	Inbound call routed by SCCS only to RAN or Music or IVR (CallPilot)
f9	CDR for telephone to telephone	0.39	CDR internal call
f10	CDR for telephone to trunk	0.32	CDR external call
f11	Internet Telephone to Virtual Trunk	0.90	Direct connection on a LAN or WAN (no VGMC involved)
Note: Only incoming calls are treated by SCCS.			

Real-time factors

Real-time factors for basic call types are mutually exclusive (for example, f1, f2, and f3). The real-time factor is calculated from the following formula:

> fi = (real-time of a featured call in milliseconds - real-time of a basic call in milliseconds) ÷ real-time of a basic call in milliseconds

Real-time factors for applications are additive (for example, f0+f2+f5, f3+f6, f0+f4+f5+f6, and so on). Real-time factors for features with similar functions (for example, RAN and IVR) are usually mutually exclusive. For a complete list of features and real-time factors, see *Large System: Planning and Engineering* (553-3021-120).

EBC rating for Succession Call Server

The rated capacity for a Succession Call Server is 35,000 EBC.

> Call Server utilization = [number of call type 1 x (1+f0+f1+...+f11) + number of call type 2 x (1+f0+f1+...+f11) +...] ÷ 32,000

> ***Note:*** The 1 preceding f0 represents the basic call. A complete feature call is a basic call plus the sum of applicable real-time factors.

Calls from a telephone that terminate on many different terminals make calculating CCS difficult. For example, Internet Telephone calls terminate on other Internet Telephones, on analog (500/2500-type) telephones, on T1 trunks, on analog trunks.

To calculate differences in terminations accurately, other parameters are needed, such as intra-office ratio, inter-office ratio, tandem call ratio, incoming/outgoing breakdown. Accounting for the above parameters make the real-time calculations difficult to complete. A simpler way to calculate CCS with reasonable accuracy is to divide all traffic by two. This is somewhat accurate because CCS on a telephone/trunk is half incoming and half outgoing.

Note: Every connection involves two terminals but appears on the processor as one call. For example, 6 CCS for each Internet Telephone means 2.5 originating calls and 2.5 terminating calls (= 6 x 100 ÷ 120 ÷ 2). It can also represent five originating calls per telephone, and no terminating call. The important point is not to double count traffic during the conversion of CCS to calls.

Calculating TLAN bandwidth for IP voice traffic

Use the following formula to calculate the bandwidth requirement for voice traffic on the TLAN:

Bandwidth in Bit/s = Erlangs on TLAN x 95 kbit/s

Note: The result of total Erlangs appears on page 180.

The calculated data rate is the incremental data bandwidth a TLAN has to reserve for the IP traffic to prevent QoS of VoIP traffic from deteriorating. The most conservative bit rate requirement is used here, which corresponds to using G.711 codec (64 kbit/s per channel) with a typical payload (20 ms). The following table shows the bandwidth required for one Erlang using a G.711 codec or G.729A codec.

Table 40
TLAN bandwidth for one channel (one Erlang)

Codec	Payload	TLAN
G.711	10 ms	126 kbit/s
G.711	20 ms	95 kbit/s
G.711	30 ms	85 kbit/s
G.729A	10 ms	70 kbit/s
G.729A	20 ms	39 kbit/s
G.729A	30 ms	29 kbit/s

See “Call Center example” on page 227 for guidance when applying engineering procedures.

WAN engineering

Specific network architecture details are required to properly engineer a Wide Area Network. For details on data networking, see *Data Networking for Voice over IP* (553-3001-160).

The Succession 1000 system can serve as a Branch Office to feed traffic over a data network to a Main Office for advanced services. The link between two offices can be a data network or traditional PSTN routes, such as PRI or TIE trunks. The data network can be LAN or WAN which is discussed here.

A Succession Call Server can route packets from Internet Telephones to a remote Main Office directly over a data network. The packet route is known as a Virtual Trunk. The Virtual Trunk traffic is an extension of the originating TLAN traffic. Virtual Trunk traffic routes to the destination node through LAN or WAN without transcoding. The Virtual Trunk requirement is calculated based on the same procedure and table used for regular trunk or PRI trunk. For details about the Branch Office VGMC traffic, see "Branch Office example" on page 215.

Data from Table 41 can be used to calculate incremental WAN bandwidth requirement to carry a one Erlang voice channel from a voice network.

Table 41
WAN bandwidth for one channel (one Erlang)

Codec	Payload	WAN
G.711	10 ms	48 kbit/s
G.711	20 ms	40 kbit/s
G.711	30 ms	37 kbit/s
G.729A	10 ms	20 kbit/s
G.729A	20 ms	12 kbit/s
G.729A	30 ms	9 kbit/s

For assumption details, refer to *IP Line: Description, Installation, and Operation* (553-3001-365).

Incremental WAN bandwidth = inter-office traffic in Erlangs x WAN kbit/s.

> ***Note:*** Using Table 41, determine WAN kbits/s based on the selected codec and payload.

Basic assumptions are that Internet Telephones generate data packets to Succession 1000 system through the TLAN. If the traffic is network traffic, that traffic can route through traditional PRI trunks or Virtual Trunks.

T1 trunks generate traffic to the VGMC and PRI, but no traffic to the WAN route. Virtual Trunks do not use a VGMC channel or PRI, but create packet data to the WAN route. If Virtual Trunk traffic uses the same TLAN to reach a router, add the traffic to the TLAN calculation. For mixed PRI and Virtual Trunk traffic, calculate for both. TLAN and WAN bandwidth requirements are incremental, since existing data network can be unknown.

Measure an existing data network to determine whether it has enough spare bandwidth to carry the additional data packets generated by the Internet Telephone voice traffic.

Traffic model

Contents

This section contains information on the following topics:

System traffic sources

The main VoIP traffic sources are the i2002 and i2004 Internet Telephones, and the i2050 Software Phone. Internet Telephones can be located in a wide geographical area that traverses through the Intranet and Internet to reach the TLAN associated with a Succession Media Gateway.

Basic telephony features are controlled and provided by the Succession Call Server. Peer networking capabilities and IP telephone calls between different networks are controlled by Meridian Customer Defined Networking (MCDN) and Q.SIG

Internet Telephones originating calls

When Internet Telephones are traffic sources, they can initiate calls to other Internet Telephones on the packet network. Internet Telephones can initiate calls through a gateway to reach traditional private network-ESN, or PSTN. An Internet Telephone to Internet Telephone call uses the signaling resource of the Succession 1000 system, but not the DS-0 channel on the gateway. An Internet Telephone to ESN/PSTN call goes through the gateway to reach the traditional voice network and occupies a DS-0 channel for the duration of the call.

Internet Telephones terminating calls

Internet Telephone to circuit-switched channels on the system

These Internet Telephone calls terminate on analog (500/2500-type) telephones, digital telephones, analog trunks, or PRI trunks. These calls generate traffic on the TLAN, VGMC, and terminating ports.

Internet Telephone to Internet Telephone

Internet Telephone to Internet Telephone calls generate signaling message to the Succession Call Server. They do not generate voice packets. If the Internet Telephones are on the same premises as the Succession 1000 system, the TLAN carries the voice packets. If the Internet Telephones are not on the same premises as the Succession 1000 system, there is traffic impact on the gateways of the originating and terminating Internet Telephones.

Internet Telephone to Virtual Trunks

This type of call is similar to an Internet Telephone to Internet Telephone call, except that the terminating telephone must be accessed through a Virtual Trunk group. This connection is subject to the maximum of 200 Virtual Trunks at any given time.

Engineer Virtual Trunks as if they are physical trunk circuits. Virtual Trunks are subject to the same traffic model (such as the Poisson formula) that define the grade of service and circuit carrying capacity of a trunk route. Virtual Trunks have normal bandwidth requirements, but deciding where to place the traffic on the data network is subject to specific network architecture.

Data network bandwidth

To engineer a Succession 1000 system in a LAN network, planners must consider the structure of existing data networks. Without knowing the end-user's network structure, Succession 1000 resources are calculated to provide the required voice services. The incremental bandwidth required on a LAN is calculated to carry the extra voice traffic served by the Succession 1000 gateway.

An incoming call to a Succession 1000 system always generates packets to the TLAN whether the call utilizes a DS0 channel in the Succession 1000 system. Therefore, take all system calls into account when calculating the TLAN traffic and bandwidth requirement.

Traffic engineering

Succession 1000 system traffic engineering is always done on an individual node basis. If the node is a part of a network, traffic between nodes is considered when calculating digital (PRI) trunking facilities.

Traffic activity

Traffic is the amount of activity a communications system generates during a given time from telephones, trunks, agents, and fax ports. All traffic is bidirectional.

Centi Call-Second and Erlang

Centi Call-Second (CCS) and Erlang are measurements of telephone traffic. CCS is 100 seconds of telephone conversation. One hour of telephone traffic equals 36 CCS or 1 Erlang.

Busy hour

For traffic engineering, the capacity of a telephony system is expressed in terms of Busy Hour. As a standard, Busy Hour is based on the average of the 10 busiest hours of the 10 busiest days of a telephony system over a three-month period. However, each end-user tends to have their own definition of Busy Hour based on their business. The end-user can define

Busy Hour as the traffic level the business wants its telephony system to be engineered for, regardless of how Busy Hour has been obtained.

Real-time engineering

The real-time engineering of a Succession 1000 system is based on the Succession Call Server's ability to meet all service requirements for processing traffic. The SSCs on other Succession Media Gateways, which are not system Succession Call Servers, provide limited processing functions and are not engineered individually. The Succession Call Server's real-time performance requirements and constraints are directly affected by the base hardware platform and the type of calls presented to it.

Traffic model

There are two types of parameters or variables to a traffic model:

- Model Input
- Model Output

Model Input

Model Input is defined as an end-user's telephone user input. If telephone user input cannot be provided, default numbers are used. End-user input or default numbers are used to calculate the level of hardware/resources required to support telephone user needs.

The end-user can provide the following parameters for engineering calculations:

- Number of i2002/i2004 Internet Telephones and CCS for each telephone.
- Number of i2050 Software Phones and CCS for each telephone.
- Sum of i2050 Software Phones plus i2002/i2004 Internet Telephones, if there is no traffic differentiation between the two.
- Number of analog terminals and CCS for each telephone or trunk.

- Percentage of traffic distribution from:
  - Internet Telephone calls to Internet Telephones
  - Internet Telephone calls to PSTN through the Succession Media Gateway
  - Internet Telephone calls to Meridian 1 or Succession 1000M private network
  - Internet Telephone calls to analog (500/2500-type) telephones and analog trunks on the Succession 1000 system
- Number of seconds average hold-time of calls (default: 180 seconds for each agent, 120 seconds for each non-agent telephone).
- G.711 or G.729A codec selection on the VGMC.
- Percent of calls involving Symposium Call Center Server (SCCS) treatment, IVR (CallPilot), and both SCCS and CallPilot.

  Note: A RAN trunk takes one card slot, but its traffic is embedded in a terminal (line or trunk), and is not specifically calculated.

Model Output

Model Output is defined as a result of the calculations made from Model Input to define the hardware and resources required to support telephone user needs. The following system resources are determined from the engineering calculations:

- Number of required VGMCs
- Number of required PRI cards (T1/E1) to PSTN
- Number of required PRI cards (T1/E1) to M1 (ESN)
- Number of required XALC for analog (500/2500-type) telephones, modems, and fax access
- Number of required XDLC for PC-based consoles and digital telephones
- Number of required analog trunk cards for analog trunks, RAN trunks, music trunks, and paging trunks
- Number of required Succession Media Gateways and Succession Media Gateway Expansions

- Amount of incremental TLAN bandwidth required to carry voice packets
- Amount of real-time Succession Call Server utilization

Note: Depending on the purpose of the engineering exercise, sometimes the input and the output of the model are reversed. The general engineering procedure should apply in either case.

Calculation formula, rules, and models

The following outlines the calculations needed to engineer a system:

- Calculate VGMC and PRI trunk requirements based on the Erlang B model (infinite sources, blocked call cleared).
- Calculate analog line card requirements (16 ports/card) and analog trunk card requirements (8 ports/card) based on port numbers.
- Calculate digital line card requirements (16 ports/card).
- Calculate number of registered non-blocking agents for each card based on VGMC port number (32).
- Calculate Digital trunk requirements based on 24 channels (T1) for each PRI card or 30 channels (E1) for each PRI card. The D-channel is included with the B-channels and is not considered separately for this calculation.
- Calculate processor capacity to the rated call capacity. The rated call capacity is based on 70% of the processor’s absolute loading.

Quality of Service (QoS)

The QoS of a network is defined by the packet loss and Round Trip Delay (RTD) of packets. These service parameters are largely determined by the network bandwidth available between communicating parties. The Call Server maintains and controls QoS statistics.

The network can provide excellent voice quality if the following QoS criteria are met:

- less than 0.5% packet loss
- less than five m/sec network RTD

The network can provide a satisfactory QoS if the following minimum service criteria are met:

- less than 5% packet loss
- less than 200 m/sec network RTD

> ***Note:*** QoS, as defined by ITU-T recommendation and simplified E-model implemented by VGMC, both use one-way transmission delay as a parameter. Any delay measurements that are round trip must be halved to be comparable to industrial standard and VGMC traffic reports.

Grade of Service (GOS)

Grade of Service (GOS) is expressed as **P**. The symbol **P** indicates the number of calls out of 100 that can be blocked. For example, P.01 means 1 call out of 100 calls can be blocked.

The GOS of a telephone system is defined by the following:

- Dial tone delay of a telephone going off-hook to make a call.
- Call blocking encountered when making a call.

In other words, both delay and blocking criteria must be met to satisfy GOS requirements. The minimum GOS for private switch/private network are:

- probability of delay (d >= three seconds) <= 0.015
- probability of blocking <= 0.01

Probability of delay (d >= three seconds) <= 0.015

The probability that there will be a dial tone delay of three seconds or more for all telephones going off-hook should be 1.5% or less. Dial tone delay is caused by one or both of the following:

- not enough processor capacity
- not enough dial tone circuits

With adequate tone circuits, the processor typically can meet the delay criterion by loading up to 70% of its absolute utilization capacity. 70% is referred to as the Rated Call Capacity of a processor. This explains why CPU utilization in some switches can be greater than 100% of the Rated Call Capacity in traffic reports. The Rated Call Capacity is the recommended real-time engineering level for Succession 1000 systems.

Probability of blocking <= 0.01

The probability of blocking implies that no more than one percent of all calls attempting to make a connection are blocked. This blocking probability of 0.01 is expressed as P.01. A GOS of P.01 is appropriate when determining channel requirements in a system.

Internal network blocking depends on the ratio of traffic-generating telephones and conversation channels. The Erlang B formula assumes that the traffic source to channel ratio is high, and the blocked calls on the first attempt are dropped.

A non-blocking configuration for a call center application on a Succession 1000 system means that a registered agent's telephone always has a TDM channel on the VGMC. The VGMC has 32 agents provisioned on the card's 32 ports. However, whether the call can reach an idle terminal or a PRI channel out of the Succession 1000 system depends on the network architecture. Non-blocking is only engineered within a Succession 1000 node. Network non-blocking engineering is outside the scope of this document.

Mixed Call Center and Main Office example

Contents

This section contains information on the following topics:

Introduction

This chapter provides a mixed call center and Main Office example for a Succession 1000 system.

In this chapter the following abbreviations are used in formulas:

- IT = Internet Telephone
- AT = analog (500/2500-type) telephone
- MC = Voice Gateway Media Card (VGMC)

Assumptions

The section provides an example of equipment and traffic characteristics for the Call Center and Main Office.

Equipment characteristics

Table 42 contains equipment characteristics for the mixed Call Center and Main Office.

Table 42
Call Center/Main Office equipment characteristics

Number	Type	Comments
300*	i2002/i2004 Internet Telephones	Used as a gateway to an ESN and a CO through PRI connections.
40	i2050 Software Phones	Used as a gateway to an ESN and a CO through PRI connections.
60*	i2002/i2004 Internet Telephones	*60 telephones from the 300 telephones above. Used as ACD agent telephones telecommuting from homes.
240*	i2002/i2004 Internet Telephones and i2050 Software Phones	*The remainder of the 300 telephones above, minus the 60 used for ACD agents. Used by workers stationed at several remote Branch Offices, using this Succession Call Server for call processing.
5	RAN trunks	Equipped to provide RAN service to PSTN and Succession 1000M and Meridian 1 users.
6	analog line ports	Used for fax.
40%	CallPilot (201i IPE version)	Provided to serve 40% of incoming calls to agents.

Traffic characteristics

The mixed Call Center and Main Office example has the following traffic characteristics.

Table 43
Call Center/Main Office traffic characteristics

Number	Comments
33 CCS	For each ACD-agent.
6 CCS	For each Office-user.
9 CCS	For each fax port. All traffic goes to ESN trunks.
0 CCS	No ACD agent traffic to and from other agents.
50%	Internet Telephone to Internet Telephone calls for office users.
40%	Non-agent Internet Telephone calls use the Succession 1000 system as a gateway to reach a PSTN network.
60%	Non-agent Internet Telephone calls terminate on a Succession 1000M and Meridian 1 (ESN) through the gateway and PRI tie trunks.
100%	All agent calls come from the PSTN.
P.01	GOS for all PRI traffic.

Required calculations

The following is a summary of the required calculations:

- Number of VGMC ports.
- Number of PRI to Succession 1000M and Meridian 1 and to PSTN.
- Number of Succession Media Gateways and Succession Media Gateway Expansions.
- Real-time load for the Succession Call Server.
- Bandwidth of the TLAN.

Mixed Call Center and Office calculation procedures

Agent telephones are registered on non-blocking VGMCs. The non-blocking VGMCs handle only agent traffic. Non-agent Internet Telephones register on other VGMCs. Agent Internet Telephones and non-agent Internet Telephones should not be mixed on the same VGMCs.

Procedure 5
Calculating traffic

1 Calculate the non-agent traffic for VGMC ports:

CCS for each IT x (1 - Intra-IT %) x (Total IT - agent IT) = Non-agent IT traffic offered to MC ports

6 x (1- 0.5) x (300 + 40 - 60) = 840 CCS.

2 Calculate the Internet Telephone to Internet Telephone (traffic not involving Internet Telephone to VGMC ports):

CCS for each IT x (Intra-IT %) x (Total IT - agent IT) = Non-agent IT traffic offered to MC ports

6 x 0.5 x (300 + 40 - 60) = 840 CCS

3 Calculate the agent traffic for Internet Telephone to VGMC ports:

Number of IT agent sets x CCS for each agent = Agent traffic to MC

60 x 33 = 1980 CCS

4 Calculate the total traffic in Erlang for TLAN bandwidth calculation:

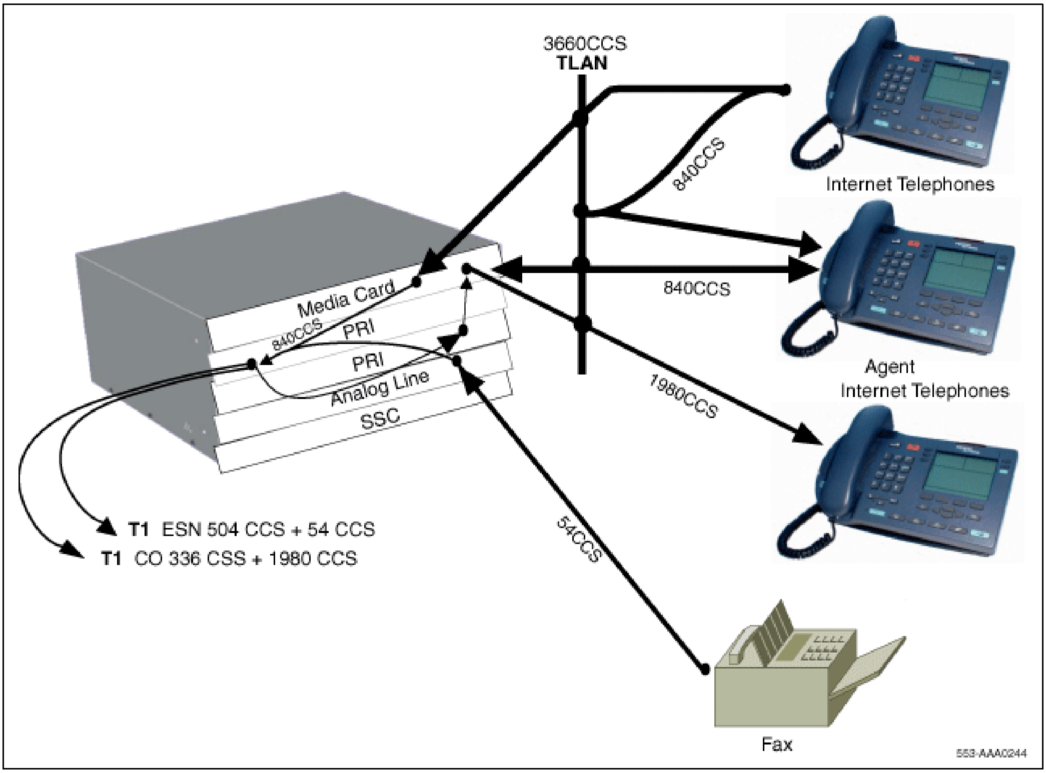
a. Non-agent traffic for MC + IT to IT non-MC + agent traffic for IT to MC = Total CCS traffic (840 + 840 + 1980 = 3660 CCS)

b. Total CCS traffic ÷ 36 = Erlangs (3660 ÷ 36 = 102 Erlangs)

End of Procedure

Figure 31 on page 209 is a conceptual representation of traffic distribution. An actual system could require a different type and number of cards.

Figure 31
Mixed office traffic flow



Procedure 6
Determining Internet Telephone to Voice Gateway Media Card

1 Calculate for the number of non-agent VGMCs:

Non-agent IT traffic ÷ MC capacity = Non-agent MC required

Note: See Table 34 on page 181 for the 32-port VGMC with P.01 GOS.

840 ÷ 794 = 1.1, round up to 2 VGMCs

2 Calculate the non-blocking agent VGMCs:

Number of agent IT ÷ MC ports = Non-blocking agent MC

60 ÷ 32 = 1.88, round up to 2 VGMCs

Requirement: Two VGMCs (four spare ports in the third card)

Note: Mixing regular Internet Telephone traffic with agent non-blocking traffic on one VGMC is generally not recommended. An exception can be made when the system is limited by physical slots and there are spare ports at the agent VGMCs. In that case, moving a number of non-agent i2004 telephones (up to the number of spare ports) to register on the partial equipped VGMC could be permitted.

End of Procedure

Procedure 7
Calculating PRI cards and analog line/trunk cards

1 Calculate the non-agent PSTN traffic:

Non-agent IT traffic x % of IT to PSTN = Non-agent PSTN T1 traffic

840 x 0.4 = 336 CCS

2 Calculate the agent PSTN traffic:

(Non-agent PSTN PRI traffic + Agent traffic from PSTN) ÷ PRI capacity at P.01 = Number of PRI cards

(336 + 1980) ÷ 550 = 4.21 PRI cards

Requirement: Five PRI cards are required to serve PSTN.

3 Calculate the Succession 1000M or Meridian 1 ESN traffic:

a. Non-agent IT trunk traffic x % IT traffic to ESN + fax traffic = ESN PRI traffic (840 x 0.6 + 9 x 6 = 558 CCS)

b. ESN traffic CCS ÷ PRI capacity at P.01= Required PRI cards (558 ÷ 550 = 1.01)

Requirement: One PRI card is required to connect to the system.

4 Calculate the total number of PRI cards:

PSTN PRI + ESN PRI = Total PRI cards

5 + 1 = 6 Total PRI cards

Requirement: Six card slots for PRI services.

Other service cards.

Requirement: Five RAN trunks need one MIRAN card. Six fax lines need one XALC card.

Requirement: Two service cards for other services.

See Table 44 to determine the chassis requirement for cards.

End of Procedure

Procedure 8
Calculating Succession Media Gateway and Succession Media Gateway Expansion

Table 44 shows the number of cards required in the Succession Media Gateway or Succession Media Gateway Expansion.

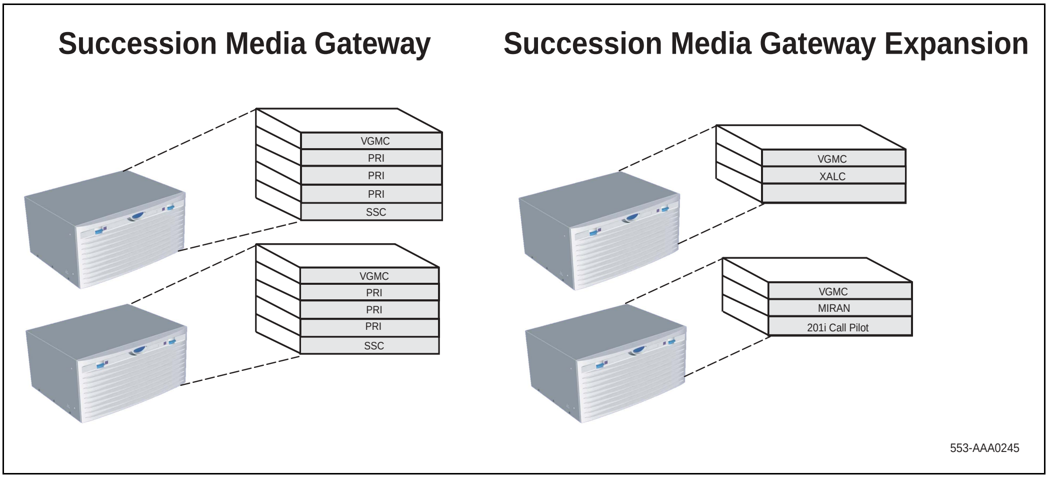
Table 44
Card quantity

Card type	Number of cards
VGMC	4 (see Note)
XALC	1
PRI	6
MIRAN	1
201i CallPilot	1
Note: Two VGMCs required for agent Internet Telephones. Two VGMCs required for other Internet Telephone traffic.	

Requirement: Two Succession Media Gateways and two Succession Media Gateway Expansions.

One possible distribution of cards are shown in Figure 32 on page 213.

Figure 32
Possible Mixed Call Center and Main Office equipment layout



End of Procedure

Procedure 9
Calculating Succession Call Server real -time

1. Calculate the number of Internet Telephone to Internet Telephone calls:

   Non-MC IT to IT traffic x 100 ÷ Average hold time in seconds ÷ 2 = IT to IT calls

   840 x 100 ÷ 120 ÷ 2 = 350 calls

2. Calculate the number of Internet Telephone to non-Internet Telephone calls through VGMC ports:

   Non-agent IT to PRI traffic x 100 ÷ Average hold time in seconds = IT non-agent to PRI calls

   840 x 100 ÷ 120 = 700 calls

3. Calculate the of number Internet Telephone agent calls with IVR:

   PRI to IT agent CCS x % IVR treatment x 100 ÷ Average hold time in seconds = IT calls with IVR

   1980 x 0.4 x 100 ÷ 120 = 660 calls

4 Calculate the number of Internet Telephone agent calls without IVR:

PRI to IT agent CCS x (1 - % IVR treatment) x 100 ÷ average hold time in seconds = IT calls not using IVR

1980 x 0.6 x 100 ÷ 120 = 990 calls

5 Calculate the number of Fax calls:

fax CCS x number fax ports x 100 ÷ 120 = fax calls

9 x 6 x 100 ÷ 120 = 45 fax calls

6 Calculate the number of Succession Call Server load:

IT to IT calls x (1+f1) + IT non-agent to PRI x (1+f2) + IT agent calls without IVR x (1+f0+f2) + IT agent calls with IVR x (1+f0+f2+f6) + fax calls x (1+f4) = Total EBC

350 x (1+1.15) +700 x (1+ 0.68) + 990 x (1+0.13+ 0.68) + 660 x (1+0.13+ 0.68 + 3.68) + 45 x (1+0.18) = 7396.9

7396.9/35,000 = 21%

The rated capacity of a Succession Call Server is 35,000 EBC. The proposed configuration uses about 21% of the CPU capacity.

End of Procedure

Procedure 10
Calculating TLAN bandwidth for Internet Telephone traffic

Additional bandwidth required at TLAN to carry voice traffic:

1 Calculate the Data rate:

Total traffic in Erlang (from page 208) x 95 kbit/s

102 x 95 = 9,690 kbit/s = 9.69 Mbit/s

All Internet Telephone calls to ESN and PSTN are carried by the PRI. There is no Virtual Trunk traffic, therefore no WAN bandwidth calculation is required.

The equation above is based on the assumption of G.711/20 ms codec and packet size, which is conservative (slightly overestimating the bandwidth requirement) if substituted for other practical combinations.

End of Procedure

Branch Office example

Contents

This section contains information on the following topics:

Introduction

This chapter provides a Branch Office or a non-call center example for a Succession 1000 system.

In this chapter, the following abbreviations are used in the formulas:

- IT = Internet Telephone
- AT = analog (500/2500-type) telephone
- MC = Voice Gateway Media Card (VGMC)

Assumptions

The Branch Office example has the following equipment and traffic characteristics.

Equipment characteristics

The Branch Office example has the following equipment.

Table 45
Equipment characteristics

Number	Type
120	Internet Telephones
36	analog (500/2500-type) telephones
6	RAN trunks

Traffic characteristics

The Branch Office example has the following traffic characteristics.

Table 46
Traffic characteristics

Number	Comments
6 CCS	For each Internet Telephone, 50% of the calls go to other Internet Telephones, and 50% of the calls go to analog (500/2500-type) telephones and PRI trunks.
5 CCS	For each analog (500/2500-type) telephone, 40% of the calls go to PRI trunks, and 60% of the calls go to Internet Telephones.
120 seconds	For the average holding time for each call.

Note: Conference traffic, like other applications, is generally not singled out for calculation in traffic engineering. When a Branch Office does not have conference capability, conference call participants must use the WAN to reach Main Office to join conferences. However, if the traffic is significant (a rough guide is more than 10% of Internet Telephone traffic), traffic should be included in WAN bandwidth calculation.

Required calculations

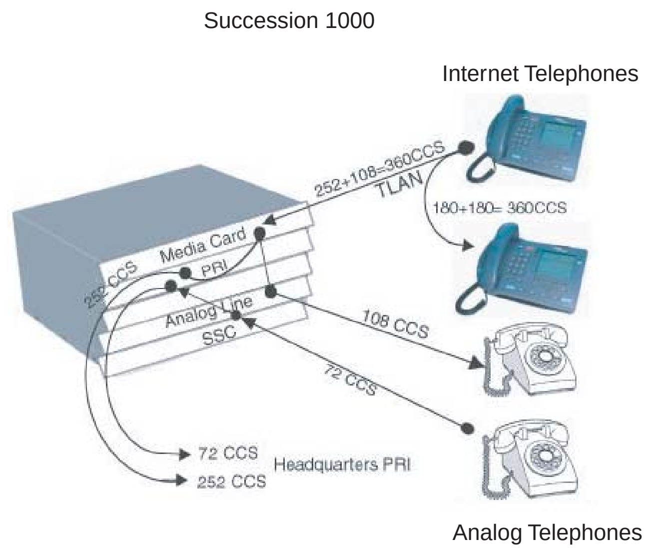
The following is a summary of the required calculations:

- number of PRI channels to the headquarters
- number of VGMCs
- number of Succession Media Gateways and Succession Media Gateway Expansions
- real-time load for the Succession Call Server
- bandwidth for LAN

Branch Office calculation procedures

Figure 33 on page 218 is a conceptual representation of Branch Office traffic flow.

Figure 33
Branch Office traffic flow



Notes:
1. Internet Telephone to Internet Telephone calls do not go through the Voice Gateway Media Card.
2. Analog telephone calls to the PRI do not go through the Voice Gateway Media Card.
3. All other calls go through the the Voice Gateway Media Card.

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Procedure 11
Calculating traffic

1 Calculate the total Internet Telephone traffic to the LAN:

a. Total IT x IT CCS = Total CCS

120 x 6 = 720 CCS

b. Total CCS ÷ 36 = Erlangs

720 ÷ 36 = 20 Erlangs

2 Calculate the VGMC traffic from Internet Telephones to PRI and analog (500/2500-type) telephones:

Total CCS x % IT – Non-IT ports = IT traffic through VGMC

720 x 0.5 = 360 CCS

Requirement: The VGMC with 794 CCS capacity handles only 360 CCS. Only one VGMC is needed.

3 Calculate the Internet Telephone to Internet Telephone traffic (Intra-Internet Telephone traffic does not go through VGMC):

Total CCS x % calls to Intra-IT calls = Non VGMC traffic

720 x 0.5 = 360 CCS

4 Calculate the analog (500/2500-type) telephone traffic to Internet Telephone:

a. Number of ATs x CCS for each AT x % of calls to ITs
= Analog traffic to IT

36 x 5 x 0.6 = 108 CCS (this traffic is part of VGMC traffic)

b. Number of ATs x CCS for each AT x % of calls to PRI
= Analog traffic to PRI

36 x 5 x 0.4 = 72 CCS (this traffic does not go to the VGMC)

5 Calculate Internet Telephone traffic to PRI:

CCS of IT traffic - analog traffic to IT = IT traffic to PRI

360 - 108 = 252 CCS

6 Calculate the Total traffic to PRI:

AT traffic to PRI + IT traffic PRI = Total traffic to PRI:

72 + 252 = 324 CCS

Requirement: Only one PRI card is needed between the Branch Office and headquarters.This PRI traffic is less than 550 CCS capacity of one PRI card.

End of Procedure

Succession Media Gateway and Succession Media Gateway Expansion requirements

Table 47 shows the number of cards required and the devices on the cards in the Succession Media Gateway or Succession Media Gateway Expansion.

Table 47
Card quantity and the devices on the card

Card type	Number of cards	Devices on cards
VGMC	1	120 Internet Telephones
XALC	3	36 analog (500/2500-type) telephones
PRI	1	1 T1
MIRAN	1	6 RAN trunks

Requirement: One Succession Media Gateway and one Succession Media Gateway Expansion.

> ***Note:*** An analog line card or digital line card has 16 ports. If 36 digital telephones are used instead of 36 analog (500/2500-type) telephones, the resulting calculation would yield the same number of cards: three XDLC cards, taking the place of three XALCs. The total number of card slots and Succession Media Gateway requirements are the same.

Procedure 12
Calculating Succession Call Server loading

Only the CCS from one terminating end of a connection is used in the Call Server loading calculation. For example, a call from one Internet Telephone to another includes the CCS from both telephones. The Succession Call Server loading calculation divides the CCS by two.

1 Calculate the number of Internet Telephone to Internet Telephone calls:

IT to IT CCS x 100 seconds ÷ average hold time ÷ 2 = IT to IT calls

360 x 100 ÷ 120 ÷ 2 = 150 calls

Note: Dividing by 2 is only required for IT to IT traffic. Intra-IT CCS is double counted in relation to the number of calls. For example, one call lasting 100 seconds appears as one CCS on the originating telephone, one call on the CPU, and one CCS on the terminating telephone. Two CCS on telephones must be divided by two to get the correct number of calls.

2 Calculate the number of Internet Telephone to PRI calls:

IT to IT CCS x 100 seconds ÷ average hold time = IT to PRI calls

252 x 100 ÷ 120 = 210 calls

3 Calculate the number of Internet Telephone to analog (500/2500-type) telephones:

IT to AT CCS x 100 seconds ÷ average hold time = IT to AT calls

108 x 100 ÷ 120 = 90 calls

4 Calculate the number of analog (500/2500-type) telephone to PRI calls:

AT to IT CCS x 100 seconds ÷ average hold time = AT to PRI calls

72 x 100 ÷ 120 = 60 calls

5 Calculate the Succession Call Server loading in EBC:

IT to IT calls x (1+f1) + IT to PRI calls x (1+f2) + IT to AT calls x (1+f3) + AT to PRI calls x (1+f4) = Succession Call Server EBC

150 x 2.15 + 210 x 1.68 + 90 x 1.48 + 60 x 1.18 = 879.3 EBC

6 Calculate the Succession Call Server loading in percent:

Succession Call Server loading ÷ CS maximum EBC = % loading

879.3 ÷ 35,000 = 3%

Requirement: The loading of this configuration for Succession Call Server is very low at 3%. This Succession Call Server can be located at the Branch Office or anywhere within the zone.

End of Procedure

Procedure 13
Calculating TLAN bandwidth for Internet Telephone traffic

Use the following to calculate the incremental bandwidth required at TLAN to carry the given voice traffic:

1 Calculate the data rate:

Total traffic in Erlangs (from page 219) x 95 kbit/s

20 x 95 kbit/s = 1900 kbit/s = 1.9 Mbit/s

Note: One Erlang of TDM 64 kbit/s channel becomes 95 kbit/s packets after G.711 codec transcoding, which adds overhead. See Table 40 on page 193 if another type of codec is used.

Requirement: TLAN bandwidth 1.9 Mbit/s

End of Procedure

Procedure 14
Calculating Branch Office with Virtual Trunk WAN

If Virtual Trunks are used between the Main Office and the Branch Office instead of PRI trunks, a WAN resource is required, assuming G.711/30 ms for the WAN.

The traffic distribution is as follows:

1 Calculate the VGMC traffic:

IT traffic to analog sets, and analog sets to Virtual Trunks use VGMC:

VGMC traffic = 108 + 72 = 180 CCS

One VGMC needed (794 CCS capacity)

2 Calculate the Virtual Trunk traffic:

PRI trunk traffic now is carried by Virtual Trunks.

Virtual Trunk traffic = 72 + 252 = 324 CCS

An equivalent of 24 port channel (capacity 550 CCS from Table 34 on page 181) is sufficient to handle Virtual Trunk traffic.

3 Calculate the incremental WAN bandwidth:

Assuming G.711 and 30 ms payload, the WAN bandwidth is:

WAN bandwidth = 324/36 x 37 kb/s (G.711/30 ms Table 41 on page 194) = 333 kb/s

Refer to Table 41 on page 194 for other types of codec or payload size.

End of Procedure

Branch Office Conference Engineering (with no local conference)

Two parties at a Branch Office using Internet Telephones call each other. They conference in a third party from the same Branch Office. The conference calls use a WAN to reach the Conference Bridge at the Main Office. Refer to Table 41 on page 194 for bandwidth requirements if the codec, payload, or both are different from what is assumed in the following Branch Office Conference scenarios.

The calculated conference WAN bandwidth is added to normal WAN requirement between the Branch Office and the Main Office for ITG trunks or Virtual Trunks.

Procedure 15
Calculating unspecified Conference traffic

When you do not have specific information about conference traffic, use the following standard ratio of conference traffic to general traffic. In Nortel Networks PBX engineering, a network group of 32 loops is comprised of 28 traffic loops, 2 Conference loops, and 2 TDS. Using the ratio of 2 to 28, the conference traffic is about 7% (rounded up from 6.7%) of total traffic. Use the default value of 7% in place of specific information about conference traffic.

1 Calculate conference traffic:

Branch Office total traffic (TCCS) = # of Internet Telephones x CCS for each telephone

Conference traffic (TCON) = TCCS x 0.07 CCS = TCCS x 0.07/36 Erlangs

2 Calculate WAN bandwidth:

a. G.729A/30 ms codec: WAN kb/s = TCON (erlangs) x 9 kb/s

b. or G.711/30 ms codec: WAN kb/s = TCON (erlangs) x 37 kb/s

End of Procedure

Procedure 16
Calculating known conference traffic

When a Branch Office is known or expected to make a significant number of conference calls, traffic statistics should be collected or estimated. Use the statistics to calculate WAN bandwidth requirements.

1 Calculate conference traffic:

Cc = conference calls/busy hour [a 6-way or 3-way conference call is counted as 6 calls or 3 calls, respectively]

Ht = Average talk time (holding time) of conference in seconds (if there is no data, use 900 seconds as a default)

TCON = (3 x # 3-way conference calls + 6 x # 6-way conference calls + ...) x Ht/100 CCS = Total Cc x Ht/3600 erlangs

2 Calculate WAN bandwidth:

G.729A/30 ms codec:

WAN kb/s = TCON (erlangs) x 9 kb/s

or

G.711/30 ms codec:

WAN kb/s = TCON (erlangs) x 37 kb/s

Use other bandwidth data from Table 41 on page 194 if codec and payload are different from above combination.

End of Procedure

Branch Office Conference Engineering (with local conference MICB card)

When a Branch Office conference is provided locally at the Branch Office, there is no need to route conference traffic to the Main Office for service. A locally provided conference generates no WAN traffic, and does not require WAN bandwidth calculation.

Procedure 17 covers only simple CallPilot voice messaging traffic for WAN bandwidth calculation.

The engineering requirement for Multimedia Processing Units (MPU), such as CallPilot, depends on the type of traffic (for example, voice, fax, and speech-recognition) and services (for example, enterprise networking, network message service) involved. The MPU requirement calculations, which require several traffic tables to cover various Grade of Service practices, do not impact WAN calculation directly and are not presented here. Refer to the *CallPilot Planning and Engineering Guide* (555-7101-101) for detailed CallPilot engineering.

To leave a voice message for a user in a Branch Office, route the incoming call to the Main Office. Similarly, when a user retrieves the voice mail message, the connection is over the WAN to the Main Office. To leave or retrieve a message, the connection requires WAN bandwidth.

Procedure 17
Calculating Branch Office traffic, and WAN bandwidth without local messaging (CallPilot) capability

The following are the default parameter values used to estimate CallPilot traffic. Specific traffic information about a site should be used if known.

1 Calculate Messaging traffic:

Average holding time of a voice message: 40 seconds
(default recommended by the CallPilot NTP)

Voice Messaging Traffic (VMT) = Voice Messaging Calls x 40/100 CCS
= Voice Messaging Calls x 40/3600 erlangs

If no information about messaging calls (leaving or retrieving a message), use the following approximation:

VMT = 10% x Total Branch Office traffic in CCS = 10% x Total Branch Office CCS traffic/36 (erlangs)

2 Calculate WAN bandwidth:

G.729A/30 ms codec:

WAN kb/s = VMT (erlangs) x 9 kb/s

or

G.711/30 ms codec:

WAN kb/s = VMT (erlangs) x 37 kb/s

End of Procedure

Call Center example

Contents

This section contains information on the following topics:

Introduction

This chapter provides a call center example for a Succession 1000 system.

In this chapter, the following abbreviations are used in the formulas:

- IT = Internet Telephone
- AT = analog (500/2500-type) telephone
- MC = Voice Gateway Media Card (VGMC)

Assumptions

This Call Center example assumes that agents handle traffic from the PSTN and ESN through PRI.

Equipment characteristics

A Call Center with agents using ESN and accessing the PSTN through PRI connections has the following equipment characteristics.

Table 48
Call Center equipment characteristics

Number	Type	Comments
150	i2004 Internet Telephones	ACD agent telephones telecommuting from homes scattered in a wide area

Traffic characteristics

The Call Center has the following traffic characteristics.

Table 49
Call Center traffic characteristics

Value	Comments
100%	Incoming calls to agents are controlled by SCCS.
50%	Incoming calls use CallPilot IVR to choose services.
33 CCS	For each ACD-agent. No agent traffic comes from other agents.
90%	Internet Telephone calls come from the PSTN, the remaining 10% of calls come from an ESN.
P.01	GOS for PRI trunks.
Note: All calls reach the gateway or node through PRI tie trunks.	

Required calculations

The following is a summary of the required calculations:

- Number of VGMC ports
- Number of PRI cards for ESN access
- PRI cards for PSTN access
- Number of Succession Media Gateways and Succession Media Gateway Expansions
- Real-time load for the Succession Call Server
- Bandwidth of the TLAN to support the voice traffic required by this site

Call Center calculation procedures

The following procedures are required to establish engineering criteria.

Procedure 18
Calculating traffic

1 Calculate the required agent traffic for VGMC ports:

ACD agents x CCS for each agent = Total CCS

150 x 33 = 4950 CCS

2 Convert CCS to Erlangs for TLAN bandwidth calculation:

Total CCS ÷ 36 = Total Erlangs

4950 ÷ 36 = 138 Erlangs

End of Procedure

Procedure 19
Determining the number of Voice Gateway Media Cards

1 Calculate the non-blocking agent ports:

ACD agents ÷ ports on MC = MC required

150 ÷ 32 = 4.7 VGMCs

Requirement: Five VGMCs (with 10 spare ports)

Note: From a traffic perspective, there is little difference between configuring spare ports on five VGMCs or configuring all 10 spare ports on the fifth VGMC. For future growth, keeping all spares on one VGMC is preferred.

End of Procedure

Procedure 20
Calculating PRI cards and analog line/trunk cards

1 Calculate the PSTN traffic:

a. Total CCS x % to PSTN = PSTN CCS

4950 x 0.9 = 4455 CCS

b. PSTN CCS ÷ capacity of a 24-port PRI card = PRI cards required

4455 ÷ 550 = 8.1 PRI cards

Requirement: Nine PRI cards to provide trunk access to PSTN.

2 Calculate the Succession 1000M or Meridian 1ESN traffic:

a. Total CCS x % to ESN = ESN CCS

4950 x 0.1 = 495 CCS

b. ESN CCS ÷ capacity of a 24-port PRI card = PRI card required

495 ÷ 550 = 0.90 PRI card

Requirement: One PRI card to provide trunk access to ESN.

3 Calculate the total number of PRI cards:

PRI cards for PSTN access + PRI cards for ESN access = Total PRI cards for the system

9 + 1 = 10 PRI cards

Requirement: Ten PRI cards for the system.

End of Procedure

Required Succession Media Gateways and Succession Media Gateway Expansions

Table 50 shows the number of required cards and card slots in the Succession Media Gateway or Succession Media Gateway Expansion.

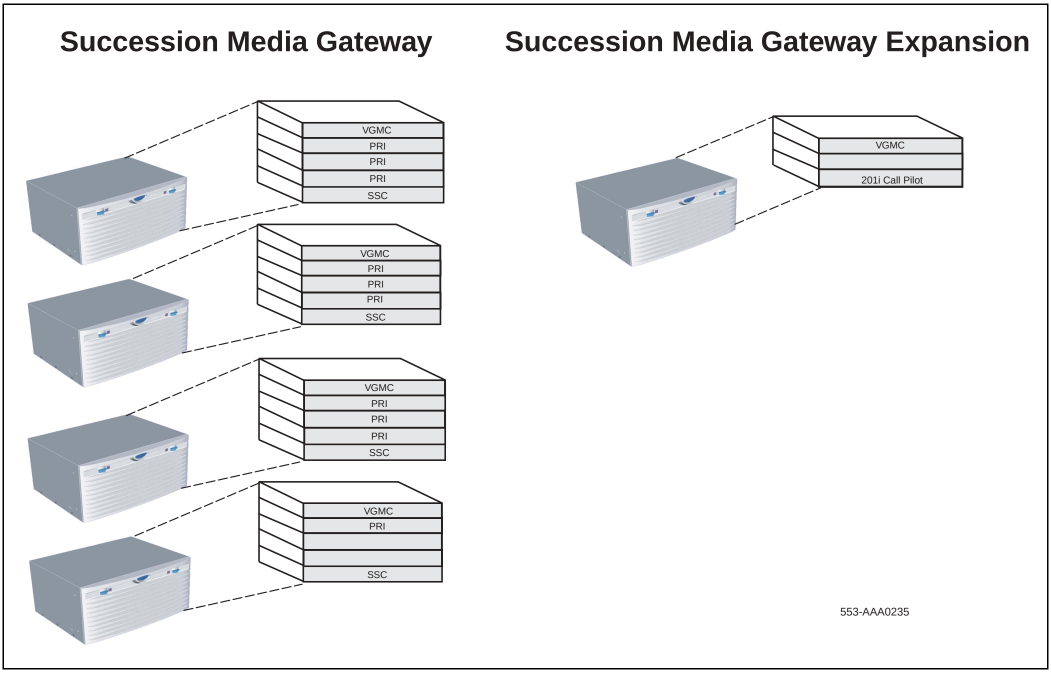
Table 50
Card quantity and required number of card slots

Card type	Number of cards	Required number of card slots	Comments
VGMC	5	5	Install in the Succession Media Gateway or Succession Media Gateway Expansion.
PRI	10	10	Install in the Succession Media Gateway (Maximum of three PRI cards for each Succession Media Gateway).
CallPilot	1	2	Install in the Succession Media Gateway Expansion.
Symposium	n/a	0	The Symposium Server is connected to Succession 1000 system through an Ethernet interface. It does not use the card slot in Succession Media Gateway or Succession Media Gateway Expansion.

Requirement: From the information in Table 50, a total of five Succession Media Gateway chassis’ and Succession Media Gateway Expansion chassis’ are required.

Figure 34 on page 232 shows a possible configuration for the Call Center hardware configuration.

Figure 34
Call Center example hardware configuration



Procedure 21
Calculating Succession Call Server real-time requirement

1 Calculate the average number of calls for each agent (assuming 180 seconds service time):

CCS for each agent x 100 seconds ÷ Service time = Average calls for each agent

33 x 100 ÷ 180 = 18.3 calls

2 Calculate the total calls receiving SCCS treatment in the system:

a. Average calls for each agent x Number of agents = Total agent calls

18.3 x 150 = 2745 calls

b. Calls x % calls receiving SCCS treatment = Total calls receiving SCCS treatment

2745 x 1 = 2745

3 Calculate the calls receiving IVR treatment:

Calls receiving SCCS treatment x % calls routed to IVR = Calls receiving IVR treatment

2745 x 0.5 = 1373 calls

4 Calculate the Real-time use in EBC:

a. Total ACD calls x (1+f0+f2) + SCCS calls x f5 + IVR calls x f6 = Real-time used in EBC

2745 x (1+0.13+0.68) + 2745 x 2.06 +1373 x 3.68 = 15676 EBC

b. Real-time used in EBC ÷ 35000 x 100 = %

15,676 ÷ 35,000 x 100 = 45%

Requirement: The rated call capacity in EBC for the Succession Call Server is 35,000 EBC. The proposed configuration uses only about 45 % of the available processor capacity.

End of Procedure

Procedure 22
Calculating LAN Bandwidth for VoIP

1 Calculate the required TLAN bandwidth to handle the given traffic:

Total ITs in Erlangs x 95 kbit/s = Bandwidth in kbit/s

138 Erlangs x 95 kbit/s = 13110 kbit/s or 13.1 Mbit/s

Note: The Erlangs used in the formula were calculated in Procedure 18 on page 229.

Requirement: The incremental bandwidth required to handle the given voice traffic on an existing LAN is 13.1 Mbit/s.

End of Procedure

Alternate Succession Call Server and survivability

Contents

This section contains information on the following topics:

Introduction

This chapter provides a brief overview of the Alternate Succession Call Server and survivability feature.

The Succession 1000 system can be provisioned with an Alternate Succession Call Server if the Succession Call Server becomes unavailable. Configure IP telephony nodes and survivable Succession Media Gateways for optimal operational efficiency and reliability.

IP telephony node configuration

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

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

> ***Note:*** The Alternate Succession Call Server is a Succession Media Gateway SSC that is configured as survivable.

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

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

If the node IDs are configured using the guidelines for the ‘Enhanced Redundancy for IP Line Nodes’ feature, then the Internet Telephones can register (if needed) to an alternate node on a Succession Media Gateway Expansion. This further improves the survivability of the Internet Telephones by allowing them to register to a different node should a system outage occur on their primary node's Succession Media Gateway.

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

Alternate Succession Call Server considerations

The following are Alternate Succession Call Server considerations:

- Succession Media Gateways are co-located.
- Configure one IP telephony node for the system (that is, all Succession Media Gateways).
- Only one Internet Telephone Connect Server (on VGMC and/or SS) is required for the node.
- Trunks in any Succession Media Gateway can be used by all users.
- Voice gateway channels (on VGMCs) in any Succession Media Gateway can be used by all users.
- Configure one survivable Succession Media Gateway as the Alternate Succession Call Server for the node.
- In Normal Mode, Internet Telephones register with the Succession Call Server.
- In Normal Mode, calls can be made between all Succession Media Gateways.
- In Survival Mode, Internet Telephones register with the Alternate Succession Call Server and can only use its resources.
- In Survival Mode, calls cannot be made between Succession Media Gateways, but all their local sets and trunks are functional.
- Less administration is required since there is only one node to manage.

Campus survivable Succession Media Gateway considerations

The following are campus survivable Succession Media Gateway considerations:

- Succession Media Gateways are in different locations.
- Configure a separate IP telephony node for each Succession Media Gateway.
- Each Succession Media Gateway requires an Internet Telephone Connect Server (on VGMC and/or Succession Signaling Server).
- At each Succession Media Gateway, provision trunks to distribute traffic and for survivability.
- At each Succession Media Gateway, provision voice gateway channels (on VGMCs).
- Configure each survivable Succession Media Gateway as the Alternate Succession Call Server for its node.
- In Normal Mode, Internet Telephones register with the Succession Call Server.
- In Normal Mode, calls can be made between all Succession Media Gateways.
- In Survival Mode, Internet Telephones register with the Alternate Succession Call Server and can only use its resources.
- In Survival Mode, calls cannot be made between Succession Media Gateways, but all their local sets and trunks are functional.
- More administration is required since there is more than one node to manage.

Survivability

A Succession Media Gateway has two modes of operation, Normal and Survival.

- In Normal Mode the local resources of a Succession Media Gateway are controlled by the Succession Call Server's call processing.
- When a Succession Media Gateway performs call processing for its local resources, the Succession Media Gateway is in Survival Mode.

Switchover to Survival Mode

A survivable Succession Media Gateway can restart after it loses communication with the Succession Call Server, due to:

- an outage of the Succession Call Server, or
- a failure of the IP link between the Succession Call Server and Succession Media Gateway.

During the restart procedure, the Succession Media Gateway attempts to register with the Succession Call Server. If a connection cannot be made with the Succession Call Server within approximately two minutes, the Succession Media Gateway switches to Survival Mode and acts as a stand-alone Succession 1000 system.

Triggers

If Survivability is configured on a Succession Media Gateway, the following two scenarios can trigger a switchover to Survival Mode:

- Automatic Switchover is triggered when the Succession Media Gateway loses communication with the Succession Call Server and the Switchover Time Out (SWOTO) timer expires. This can occur if there is a catastrophic failure of the Succession Call Server, or the IP link is lost between the Succession Call Server and the Succession Media Gateway.
- Manual Switchover is triggered with the Switchover to Survival (SOTS) command using LD 135 in *Software Input/Output: Maintenance* (553-3001-511).

Automatic Switchover to Survival Mode

When a Succession Media Gateway, with survivability configured, loses communication with the Succession Call Server, the Succession Media Gateway automatically switches over to Survival Mode when the SWOTO timer expires.

If the IP link is detected as down again before the expiration of the SWOTO timer, the timer stops, and the Succession Media Gateway remains in Survival operating mode.

The state of communication between the Succession Call Server and the Succession Media Gateway is monitored by a simple polling mechanism called a Heartbeat.

The following example illustrates the tasks performed by a Succession Media Gateway when communication with the Succession Call Server is lost.

1. The Succession Media Gateway attempts to re-establish the connection to the Succession Call Server. After four re-connection attempts with a pre-defined delay between each attempt, the SWOTO timer starts.
2. The SWOTO timer expires after the time defined with LD 117 (in *Software Input/Output: Maintenance* (553-3001-511)) elapses.
3. The Succession Media Gateway re-starts. As the Succession Media Gateway is going through the re-start procedure, it attempts to register with the Succession Call Server.
4. If a connection cannot be made to the Succession Call Server, the Succession Media Gateway comes up in Survival Mode.

Manual Switchover to Survival Mode

Manual commands are provided to allow a technician to force a switchover to Survival Mode. These commands are only available on the Succession Call Server. They can be used only if there is an established IP link between the Succession Call Server and a Succession Media Gateway. To manually switchover to Survival Mode, use the SOTS command using LD 135 in *Software Input/Output: Maintenance* (553-3001-511).

CAUTION
Service Interruption
A manually invoked switchover causes a restart of the Succession Media Gateway.

After the SOTS command has been successfully executed, the Succession Media Gateway remains in Survival Mode until the Switch Back From Survival (SBFS) command is issued in LD 135 by the technician.

CAUTION
System Failure
If the software is upgraded on the Succession Call Server, it must also be upgraded on the Succession Media Gateway in order for survivability to function.

In Survival Mode, a valid database must be downloaded to the Succession Media Gateway in order to function. The database is downloaded or ‘synchronized’ each time a datadump is performed. A carbon copy of the database on the Succession Call Server is downloaded to the Succession Media Gateway with every data dump.

Switchback from Survival Mode

A Succession Media Gateway can switch back to Normal Mode after communication with the Succession Call Server is restored.

The following two scenarios can trigger a chassis in Survival Mode to return to Normal Mode.

- Automatic Switch Back (AUTOSB) allows a Succession Media Gateway to automatically switch back from Survival Mode to Normal Mode as soon as the IP link with the Succession Call Server is restored, and the SWOTO timer has expired. A restart is initiated on the Succession Media Gateway. At the end of the system start, the Succession Media Gateway is ready to operate in Normal Mode.
- Manual Switch Back allows a technician to force the system into Normal Mode by issuing the SBFS command. This command returns the system to Normal Mode after the SOTS command has been used.

Automatic Switch Back from Survival Mode

When the Automatic Switch Back option is configured for a Survivable Succession Media Gateway, the Succession Media Gateway automatically switches back from Survival Mode to Normal Mode. This occurs as soon as the IP link with the Succession Call Server is restored and the SWOTO timer expires.

A valid database is required for the Succession Media Gateway for survivability.

The AUTOSB command is available in LD 117.

LD 117 – AUTOSB command

Command	Description
CHG AUTOSB <cab#> <Switchback setting>	cab# = 1-4, Succession Media Gateway Switchback setting = (YES) NO

When the switchback parameter is set to YES (the default), the Succession Media Gateway automatically switches back from Survival Mode as soon as the SWOTO timer expires. If switchback is set to NO, the Succession Media Gateway remains in Survival Mode until a technician enters the SBFS command.

Switchover Timer

The timer is started on a Survivable Media Gateway as soon as the IP link with the Succession Call Server goes up or down. When the timer expires, the switchover (or switch back) is triggered. The timer is used to avoid instability in the Operating Mode of the Succession Media Gateway when the IP link with the Succession Call Server becomes unstable.

The switchover timer is also used during the start-up of a Survivable Succession Media Gateway. This allows the Succession Media Gateway to go into Survival Mode if the Succession Media Gateway cannot connect to the Succession Call Server on system start-up.

When the IP link restores (for a Succession Media Gateway in Survival Mode with AUTOSB configured), the SWOTO timer is started.

If the timer expires, a switch back is initiated to change from Survival Mode to Normal Mode. If the IP link is detected as down again before the expiration of the SWOTO timer, the timer stops, and the Succession Media Gateway remains in Survival operating mode.

Manual Switch back from Survival Mode

After the SOTS command has been successfully executed, the Succession Media Gateway remains in Survival Mode until the technician issues the Switch Back From Survival (SBFS) command in LD 135.

LOCK and UNLOCK commands

The LOCK and UNLOCK commands are available from the Succession Call Server.

The LOCK command locks a Succession Media Gateway in the mode that it is in when the command is invoked. This does not require a restart of the selected Succession Media Gateway.

- If the Succession Media Gateway receives the LOCK command in Normal Mode, it goes into Normal Locked Mode.
- If the Succession Media Gateway receives the LOCK command in Survival Mode, it goes into Survival Locked Mode.
- When a Succession Media Gateway is in Normal Locked or in Survival Locked Mode, no switchover (or switch back) is possible (automatic or manual) until the UNLOCK command is issued.

The UNLOCK command unlocks a Succession Media Gateway. This does not require a restart of the selected Succession Media Gateway.

- If the UNLOCK command is received in Normal Locked Mode, the Succession Media Gateway returns to Normal Mode.
- If the UNLOCK command is received in Survival Locked Mode, the Succession Media Gateway returns to Survival Mode.
- This command has no impact if it is received by a Succession Media Gateway that has not been locked into either Normal or Survival Mode.

The LOCK/UNLOCK command can be used in any mode by a technician to keep a Succession Media Gateway in the current mode, regardless of the state of the IP link to the Succession Call Server. For example, a technician can issue a SOTS command. This forces the selected Succession Media Gateway into Survival Mode prior to restarting the Succession Call Server. A LOCK command can be issued from the Succession Call Server prior to a restart. This keeps the selected Succession Media Gateway in Survival Mode, until manually returned to Normal Mode. Manually returning to Normal Mode reboots the Succession Media Gateway.

These commands are applicable to both modes and can be used to keep a Succession Media Gateway in Survival Mode after the Automatic Switch Back occurs.

Survivability notification

Special dial tone

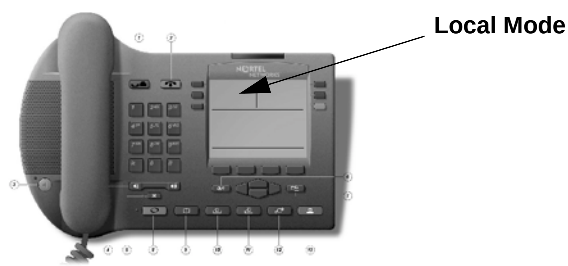
The dial tone provided to the telephones in Survival Mode is different from the dial tone for telephones in Normal Mode. This dial tone is not user configured. A fixed value is used for the cadence corresponding to the special dial tone.

Users must disable the dial tone detection option on their fax and modem for use in Survival Mode. Special dial tone can impact faxes and modems.

Display information

During Survival Mode, the Internet Telephones display “Local mode” in the first line of the display. Refer to Figure 35.

Figure 35
Telephone display



Remote TTY

Survivable IP Remote TTY

The three Serial Data Interface (SDI) ports on the Succession System Controller (SSC) of a Succession Media Gateway are available for use as additional system remote TTYs.

If the Succession Media Gateway is configured for Survivability, the SDI ports of the Succession Media Gateway SSC can be used during Survival Mode. In this mode, they function as a TTY connected to a stand-alone Succession 1000 system. However, the TTY has no access to either LD 43 or LD 143 in *Software Input/Output: Maintenance* (553-3001-511). When in Survival Mode, the SDI ports of the Succession Media Gateway cannot be used to access the Succession Call Server. Refer to Table 51 for SDI port numbering.

Table 51
SDI port numbering

Chassis	Normal Mode	Survival Mode
Succession Call Server	0, 1, 2	n/a
Succession Media Gateway # 1	3, 4, 5	0, 1, 2
Succession Media Gateway # 2	6, 7, 8	0, 1, 2
Succession Media Gateway # 3	9, 10, 11	0, 1, 2
Succession Media Gateway # 4	12, 13, 14	0, 1, 2

Special text, which is displayed on the TTY of the Succession Media Gateway, indicates when the Succession Media Gateway is operating in Survival Mode. This text informs the technician of the difference between the Succession Media Gateway TTY’s access to the Succession Media Gateway in Normal Mode and to the Succession Media Gateway in Survival Mode.

The text displayed on the TTY of the Succession Media Gateway prior to login is as follows:

```
OVL111 0000 IDLE
TTY 00 SCH MTC BUG 21:44
SURVIVAL MODE
```

The text displayed on the TTY of the Succession Media Gateway after login is as follows:

```
TTY 00 LOGGED IN 21:44 3/10/1999 SURVIVAL MODE
OVL000 SURVIVAL MODE
```

TN mapping during Survival Mode

TN mapping remains the same for both Normal and Survival Modes. For example, the second card slot of the first Succession Media Gateway is always card number 12.

Configuring trunks for survivability

To receive incoming calls or make outgoing calls in Survival Mode, the Direct Inward Dial (DID)/Direct Outward Dial (DOD) trunks must be configured on that Succession Media Gateway.

Applications/hardware on a Succession 1000 system with survivability

The following applications and hardware installed in the Succession Media Gateway are available during Survival Mode:

- IP telephony node components (Internet Telephones, VGMCs, Succession Signaling Servers)
- Analog line cards, digital line cards, and trunk cards
- Attendant Console, telephones, and trunks
- MIXX portfolio (MICB, MICA, MIPCD, MIRAN, and MIVS)
- MDECT
- Remote Office

Note: Only products/applications within the particular Succession Media Gateway function in Survival Mode.

Peripheral Software Download

Peripheral Software Download (PSDL) is available from the Succession Call Server in Normal Mode only. PSDL is not supported on Succession Media Gateways when they are in Survival Mode.

Incremental Software Management

The default Incremental Software Management (ISM) parameter for survivability allows for one Succession Media Gateway to be configured as survivable. For additional Succession Media Gateways, survivability is an orderable option. The Survivability parameter is keycode activated.

Database synchronization

Database synchronization between the Succession Call Server and a Succession Media Gateway occurs with a data dump from the Succession Call Server, or with a Succession Media Gateway start-up in Normal Mode.

During Survival Mode, Survivable Succession Media Gateways use a copy of the database that was configured at the Succession Call Server and previously downloaded to the Survivable Succession Media Gateway. The database synchronization occurs automatically upon system start-up of the Succession Media Gateway in Normal Mode or users can use the Invoke Datadump Program (EDD) command at the Succession Call Server.

Data can be changed on the Succession Media Gateway while in Survival Mode, but new or changed data is lost when switched back to the Succession Call Server. The local datadump (EDD) is supported only on the Succession Call Server.

Data dump enhancements and new commands

With the introduction of Survivable Succession Media Gateways, the EDD command is enhanced to first perform an EDD on the Succession Call Server and then download the database files to each Survivable Succession Media Gateway.

Before the file transfer, the Succession Call Server verifies that the software release of the Survivable Succession Media Gateway matches its own software release. If this check fails, the download operation is aborted for that Succession Media Gateway.

The data files are loaded from the Succession Media Gateway’s primary drive (c:) to protected memory when a switchover to Survival Mode occurs. The database files downloaded from the Succession Call Server are used only when the Survivable Media Gateway switches to Survival Mode.

Retrieving CDR records from a survivable Succession Media Gateway

Call Detail Recording (CDR) records are available from a Succession Media Gateway that has entered into Survival Mode. These records must be manually retrieved when the Succession Media Gateway returns to Normal Mode (from survival Mode).

CDR records are only generated from the Survivable Succession Media Gateway when the system is operating in Survival Mode. The CDR files are deleted from the Succession Media Gateway after a successful transfer to a PC.

Succession 1000

Succession 1000 System

Planning and Engineering

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