
Succession 1000

Succession 3.0 Software

Succession 1000 System

Overview

Document Number: 553-3031-010

Document Release: Standard 1.00

Date: October 2003

Copyright © 2003 Nortel Networks

All Rights Reserved

Produced in Canada

Information is subject to change without notice. Nortel Networks reserves the right to make changes in design or components as progress in engineering and manufacturing may warrant. This equipment has been tested and found to comply with the limits for a Class A digital device pursuant to Part 15 of the FCC rules, and the radio interference regulations of Industry Canada. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at their own expense.

SL-1, Meridian 1, and Succession are trademarks of Nortel Networks.

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: System Overview (553-3023-010).

Contents

About this document	9
Subject	9
Applicable systems	9
Intended audience	9
Conventions	10
Related information	10
Introduction	13
Contents	13
Overview	13
Nortel Networks' Enterprise VoIP strategy	14
Architecture	17
Call Server	18
Signaling Server	18
Media Gateway	18
Infrastructure	19
Applications	19
Desktop Clients	19
802.11 Wireless LAN	20
Wireless	20
System management	21
Optivity Telephony Manager	21
Simplified Web-Based management	22
IP Peer Networking (H.323 Signaling Gateway)	22
Deployment options	23

Single campus deployment	24
Multiple buildings in a campus	26
Multiple sites	27
Distributed Call Servers	29
Capabilities	30
System features	30
Power over LAN	30
Desktop clients and accessories	30
Internet Telephones	30
i2002 Internet Telephone	31
i2004 Internet Telephone	31
i2050 SoftPhone	32
Digital telephones	33
Analog telephones and devices	34
IP Adapter	34
Attendant consoles	34
Applications	35
Interworking/Interoperability	38
Migrating to an IP telephony network	38
Planning for system deployment	41
Contents	41
Planning for Succession 1000 deployment	41
Evaluate existing telephony infrastructure	41
Evaluate existing data infrastructure	42
Knowledge requirements	44
Succession 1000 numbering plans and call routing	45
Zoning plan	46
Numbering plan options	46
System component description	51
Contents	51
Succession 1000 system LAN connections	52
Succession 1000 system components	53
NTDU08 Call Server	55

NTDU27 Signaling Server	64
Signaling Server software applications	67
NTDU14 Media Gateway	73
NTDU15 Media Gateway Expansion chassis	80
Ethernet switch (customer-supplied)	83
Power over LAN (optional)	83
Internet Telephones	86
NTVQ01 Succession Media Card	88
IP Line 3.0 application	91
Software architecture	92
Call Server and Media Gateway software	92
Signaling software	92
Voice Gateway Media Card loadware	92
i2002 and i2004 Internet Telephone firmware	93
i2050 Internet Telephone application	93
Software delivery	93
Centralized Automatic Software Upgrade	93
Centralized upgrade	94
Centralized patching	94
File uploading	94
Patching implementation	94
System management	94
Element Manager	94
Call Server management	95
Media Gateway management	96
Signaling Server management	96
Voice Gateway Media Card management	97
Call Server configuration	97
Reliability strategies	99
Contents	99
Overview	100
Component redundancy	100
Call Server redundancy	101
Alternate Call Server	101

- Signaling Server redundancy 101
 - H.323 Gatekeeper redundancy 102
- Campus distributed Media Gateway in Survival Mode 104

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 describes Succession 1000 system architecture, software and hardware requirements, components, and network connections, as well as the use of a web interface to configure and maintain certain aspects of the system.

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

This document is intended for individuals responsible for configuring a Succession 1000 system.

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)
- *IP Peer Networking* (553-3001-213)
- *Branch Office* (553-3001-214)
- *Features and Services* (553-3001-306)
- *Software Input/Output: Administration* (553-3001-311)
- *Meridian Integrated RAN: Description, Installation, and Operation* (553-3001-360)
- *Meridian Integrated Call Assistant: Engineering, Installation, Administration, and Maintenance* (553-3001-362)
- *IP Line: Description, Installation, and Operation* (553-3001-365)
- *802.11 Wireless IP Gateway* (553-3001-366)
- *Telephones and Consoles: Description* (553-3001-367)
- *Internet Terminals: Description* (553-3001-368)
- *DECT: Description, Planning, Installation, and Operation* (553-3001-370)
- *Software Input/Output: Maintenance* (553-3001-511)
- *Succession 1000 System: Planning and Engineering* (553-3031-120)
- *Succession 1000 System: Installation and Configuration* (553-3031-210)

Online

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

<http://www.nortelnetworks.com/>

CD-ROM

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

Introduction

Contents

This section contains information on the following topics:

Overview	13
Nortel Networks' Enterprise VoIP strategy	14
Architecture	17
Deployment options	23
Capabilities	30
Desktop clients and accessories	30

Overview

Nortel Networks' Meridian 1 business communications system has long been the choice of large and small businesses globally, because of its reliability and wide range of features.

Contributing to this popularity is Nortel Networks' Evergreen policy, a commitment to evolve the product rather than cause users to forego major investments in order to upgrade.

The Succession 1000 system utilizes the existing communications infrastructure, and leverages the knowledge and experience of those who buy, install, and maintain voice communications. At the same time, it continues to evolve to stay with current business and technology trends.

Nortel Networks introduced Succession 1000 to address the increasing demand for Voice over IP (VoIP) and support the convergence of voice and data networks.

Succession 1000 is much more than an IP PBX. It is the architecture that evolves, using open standards, to deliver future Advanced Applications, Interactive Multimedia Services, Integrated Desktops, and Management Services.

It is the platform with which Nortel Networks introduces the evolutionary architecture and building blocks necessary for the IP future.

Nortel Networks' Enterprise VoIP strategy

Succession 1000 expands the system's capabilities to leverage the flexibility of IP WANs. It provides seamless network integration, simplified management, greater flexibility in network deployment, and reduced costs for supporting an increasingly distributed global user community.

Enterprise communications has become one of the most technologically dynamic areas of the telecommunications industry. With the emergence of VoIP and the high expectations of cost reduction, new service and application availability, and widespread interoperability, the interest and expectation for this market space remains high.

Despite the high hopes, enterprise end-users continue to have widely varying budgets, technology adoption rates, motivational drivers and other requirements for moving to VoIP.

Innovations such as the IP PBX are still in the early adoption phase. Nortel Networks is working with Enterprise Internet Telephony managers to reduce the complexity.

The truth is that communications network decisions are complex and there is no one-size-fits-all answer to meet all of the considered key requirements.

Nortel Networks understands Enterprise communication needs. We believe our solutions portfolio offers the widest array of voice, data, and converged solutions to meet today's varying market requirements.

While end-users are extremely interested in VoIP solutions, they also want stable, secure, reliable systems deployed with relatively low risk. They want investment protection and assurances that they get a solid return on their investment.

End-users want a complete feature set with value-added applications that meets the needs of their business operations today and offers productivity enhancements for their employees. And last but not least, they want all of this with comprehensive, yet intuitive, management tools that simplify network support.

To summarize, end-users want it all – reliability, innovation, return on investment, and management simplicity.

With Nortel Network's solutions, the move to VoIP is not an "all or nothing" proposition, often the reality from other industry vendors.

For those end-users who are ready, we have pure IP solutions. For end-users looking to evolve their existing networks over time, Nortel Networks offers IP-enabled solutions that offer significant investment protection and a solid migration strategy.

For those end-users who are not quite ready for VoIP, our market-leading traditional voice solutions offer rock solid solutions today, with the opportunity to add a wealth of VoIP capabilities should they ever decide to pursue VoIP in the future.

Most importantly, all this flexibility comes with the quality, security, scalability, reliability and feature-richness end-users expect from Nortel Networks.

As a key component of Nortel Networks broad Succession for Enterprise portfolio of VoIP solutions, Succession 1000 is a fully-distributed IP PBX. It supports a full spectrum of industry-leading applications and features combined with business-grade reliability, investment protection, and global availability and support.

Originally introduced to the market in mid 2001, Nortel Networks' Succession 1000 is a feature-rich and reliable IP PBX system for enterprise end-users. With Succession 1000, Nortel Networks is delivering the power of IP to enterprises worldwide.

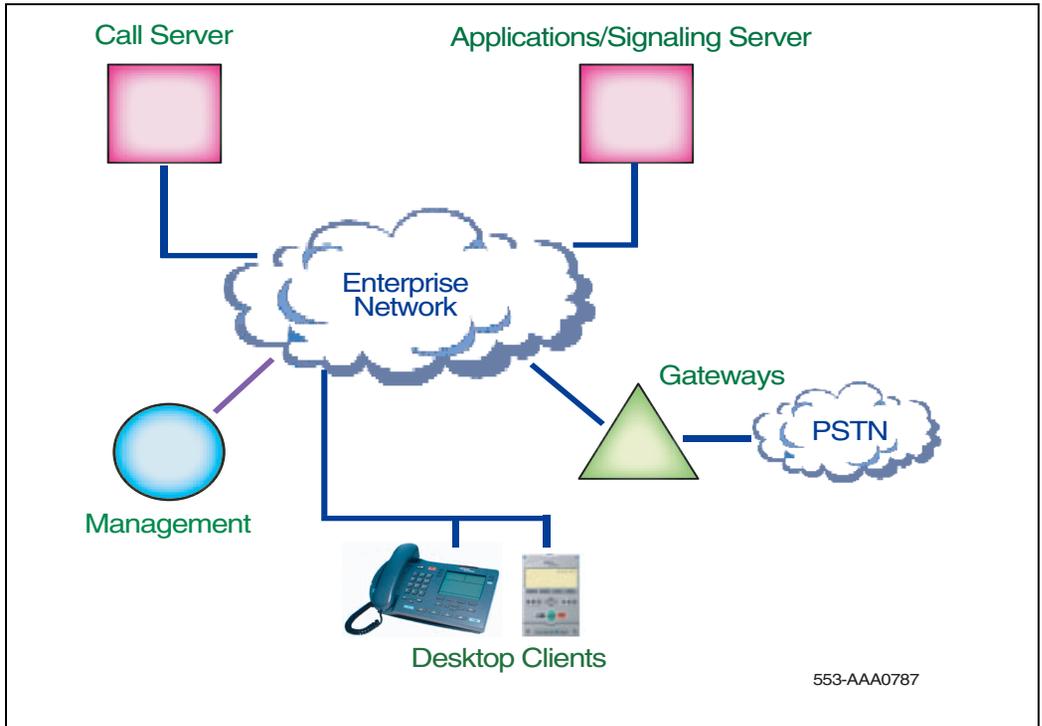
Nortel Networks is committed to providing the product, information, tools, and resources to enable our partners to succeed.

Now is the time for partners to take advantage of the tremendous growth in the VoIP market and position themselves as a complete solutions provider with the full range of Succession for Enterprise portfolio of products: Meridian 1 Internet Enabled, Business Communications Manager and Succession 1000.

Architecture

Figure 1 shows the main components of the Succession 1000 architecture.

Figure 1
Main components



Call Server

The Call Server provides telephony services, such as call processing, and supports trunking features. It also acts as a database server for synchronization of configuration information with all Media Gateways.

Signaling Server

The Signaling Server is equipped with the following software and hardware components:

- Terminal Proxy Server (TPS) software
- H.323 Gateway signaling software (virtual trunks)
- H.323 Gatekeeper software, including Gatekeeper Element Manager System Element Manager web server application

Media Gateway

The Media Gateway is a chassis that holds IPE cards and application cards. In a normal operational state, the Call Server controls the Media Gateway. The Media Gateway processor provides low-level control of the interface cards installed in the Media Gateway slots, and communicates with the Call Server for feature operation.

Note: You can configure the Succession System Controller (SSC) in the Media Gateway for survivability to assume local call processing if the Call Server cannot be accessed. If the Media Gateway is not configured as survivable, then the Media Gateway is out of service until the Call Server is again accessible.

Voice Gateway Media Cards (VGMC) are installed in the Media Gateway chassis. A VGMC provides transcoding between Internet Protocol and circuit-switched protocol.

Infrastructure

Various infrastructure components are required to support VoIP. These components include switches, routers, and Media Gateways. The data network infrastructure's engineering and provisioning is critical to achieve satisfactory telephony voice quality. For more information, see *Data Networking for Voice over IP* (553-3001-160).

Applications

Succession 1000 supports a broad suite of applications, including the following:

- CallPilot
- MIXX portfolio
 - Meridian/Succession Integrated Conference Bridge (MICB)
 - Meridian/Succession Integrated Call Assistant (MICA)
 - Meridian/Succession Integrated Personal Call Director (MIPCD)
 - Meridian/Succession Integrated Recorded Announcement (MIRAN)
 - Meridian/Succession Integrated Voice Services (MIVS)
 - Meridian/Succession DECT (MDECT) wireless telephony
- Remote Office
- Optivity Telephony Manager (OTM)
- Symposium

Desktop Clients

The i2002 Internet Telephone, i2004 Internet Telephone and the i2050 SoftPhone provide the desktop clients for Succession 1000 IP Telephony. The functionality and call features of these Internet Telephones are similar to that of a standard digital set, such as the M2616.

The Succession 1000 also supports the following telephones/terminals:

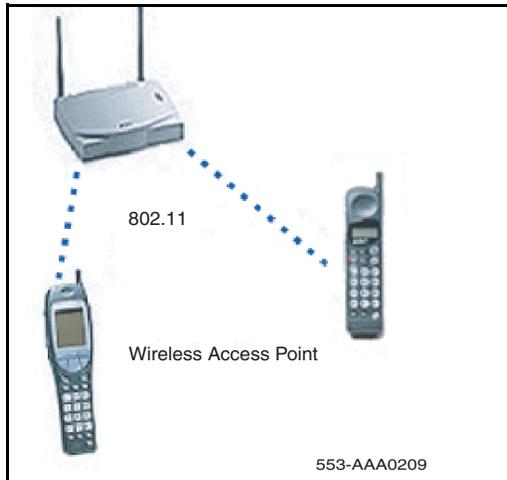
- analog (500/2500-type) telephones

- digital telephones
- T.38 Fax
- Attendant Console
- wireless telephones

802.11 Wireless LAN

The 802.11 wireless LAN is shown in Figure 2. This LAN requires an H.323 Symbol Wireless client terminal.

Figure 2
802.11 Wireless LAN



Wireless

Succession 1000 supports 802.11 Wireless IP Gateway and Digitally Enhanced Cordless Telephone (DECT) systems.

802.11 Wireless IP Gateway

The e-mobility Meridian Gateway supports communication between the circuit-switched telephony network and the H.323 Wireless IP terminals on a customer's corporate IP network. For further information, see *802.11 Wireless IP Gateway* (553-3001-366).

Digital Enhanced Cordless Telecommunications

The Digitally Enhanced Cordless Telecommunications (DECT) system is an application that enables users to move freely about their work sites using wireless handsets, and keep in communication with end-users and suppliers. Missed calls are reduced and employee productivity is increased due to quick employee response times and multi-tasking flexibility. For further information, see *DECT: Description, Planning, Installation, and Operation* (553-3001-370).

System management

Succession 1000 supports a suite of value-added management capabilities for multiple systems that reduces a customer's total cost of ownership. These include Optivity Telephony Manager and simplified web-based management.

Optivity Telephony Manager

Optivity Telephony Manager (OTM) provides management simplicity and flexible control. OTM's features include the following:

- LDAP-based directory integration
- Station Administration
- Call Accounting
- Call Tracking
- Traffic Analysis
- Maintenance
- Alarm Management
- Centralized management
- Multi-user capability

- Web-based functionality
- Customizable reporting
- Import/export utility
- Scheduled tasks
- Disaster Recovery tools

These features save time and facilities costs, simplifying management of a complex network.

Simplified Web-Based management

Succession 1000 provides a GUI-interfaced Element Manager web server that presents a web-based alternative to some of the traditional overlays and CLIs. The web server simplifies overall management with functionalities such as Gatekeeper / IP Services, IP Peer configuration, software / firmware downloads, and patch downloads.

IP Peer Networking (H.323 Signaling Gateway)

IP Peer Networking enables direct IP connections between Internet Telephones located on different systems and connected through H.323 virtual trunks. The Meridian Customer Defined Networking (MCDN) features are supported over the H.323 virtual trunks. MCDN enables features such as Calling Line Identification (CLID) to provide CDR billing capabilities and other networking features.

The H.323 Signaling Gateway must register with the H.323 Gatekeeper. The H.323 Gatekeeper manages a centralized numbering plan for the network, enabling simplified management of the Succession 1000 network.

The H.323 Signaling Gateway application that runs on the Signaling Server provides signaling support for IP Peer Networking virtual trunks. H.323 Signaling Gateway enables the system to communicate with other Succession 1000 systems or third-party H.323 gateways over an IP data network.

IP Peer Networking provides a direct media path IP connection between Internet Telephones located on different systems connected with H.323 trunks.

Because virtual trunking implies virtual switching, no circuit switching resources are used on the system with connections that involve two IP devices. Virtual trunking results in increased scalability of the system in the overall number of line and trunks supported.

Deployment options

The IP-distributed architecture of Succession 1000 offers the benefits of location flexibility for system components. The architecture also provides wiring control, and simplification of space planning, system management and databases.

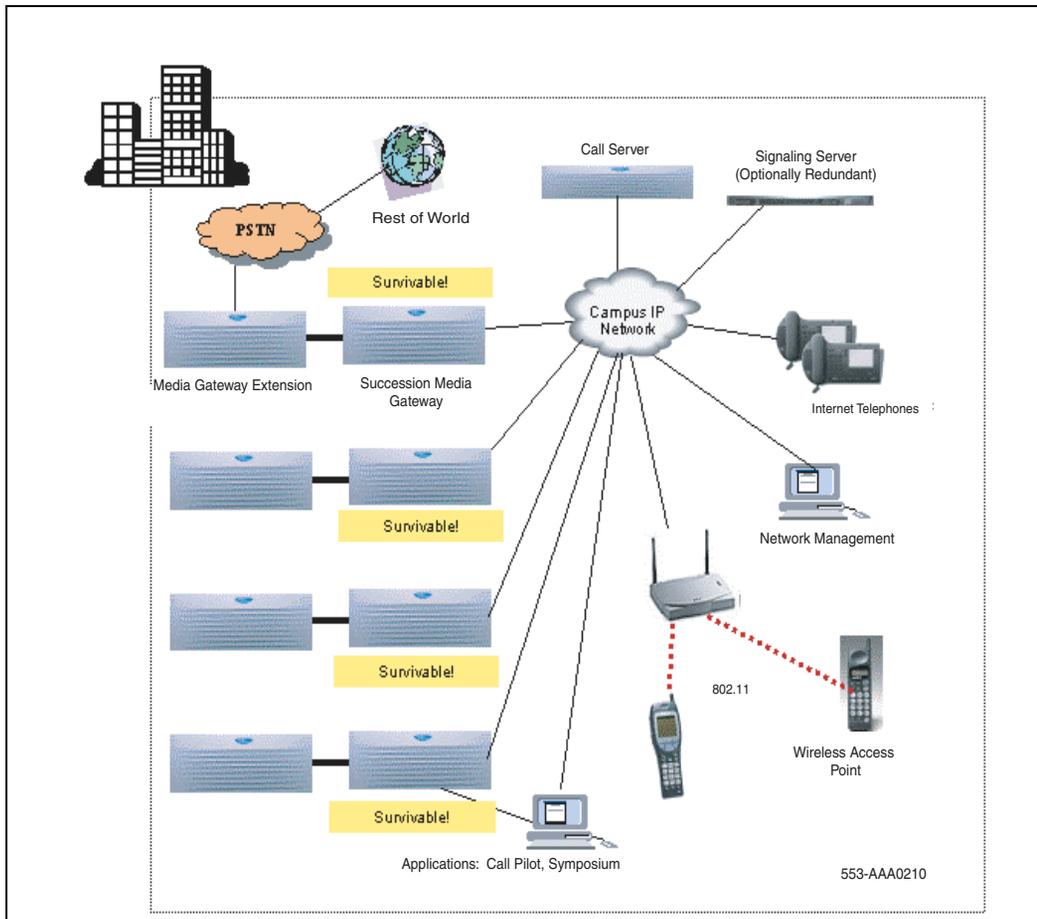
Succession 1000 bridges the circuit-switched PBX and the IP network, or can operate in an IP-only network. This consolidation of network traffic simplifies the task of managing communications to a single efficient IP network. End-users can deploy the Succession 1000 in many flexible configurations in a LAN or WAN environment. Although different combinations are possible, most installations fall into one of the following categories:

- **Single Site**
 - single building
 - multiple buildings in a campus
- **Multiple Sites**
 - central Call Server with Branch Offices users
 - distributed Call Servers

Single campus deployment

An example of a single campus deployment and its components is shown in Figure 3.

Figure 3
Single campus deployments example



Note: You can configure the network for redundancy.

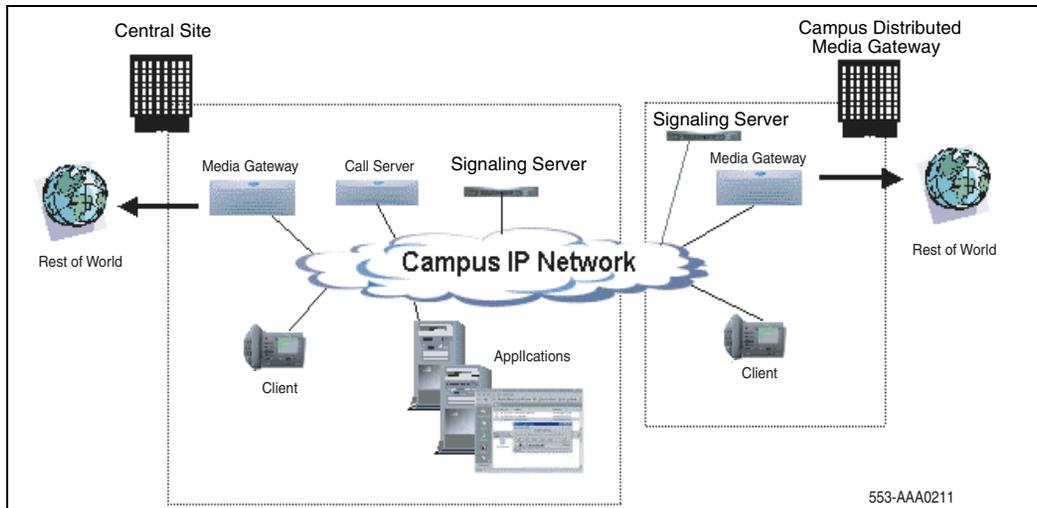
The single campus deployment example consists of the following:

- Call Server
- Signaling Server (optionally redundant)
- up to four Media Gateways and four Media Gateway Expansions
 - can be configured to provide network redundancy (see note on page 18)
 - provides a global suite of telephony trunks
 - analog telephones (service for 2500 series telephones and fax)
 - digital telephones (service for Meridian 3900 telephones)
- a switched Ethernet infrastructure
- up to 1000 Internet Telephones
- centralized management capability
- other telephony applications

Multiple buildings in a campus

Figure 4 shows a Succession 1000 system distributed across multiple buildings in a campus setting. You can distribute Media Gateways and users across a campus IP Network.

Figure 4
Multi-site deployment with centralized call processing



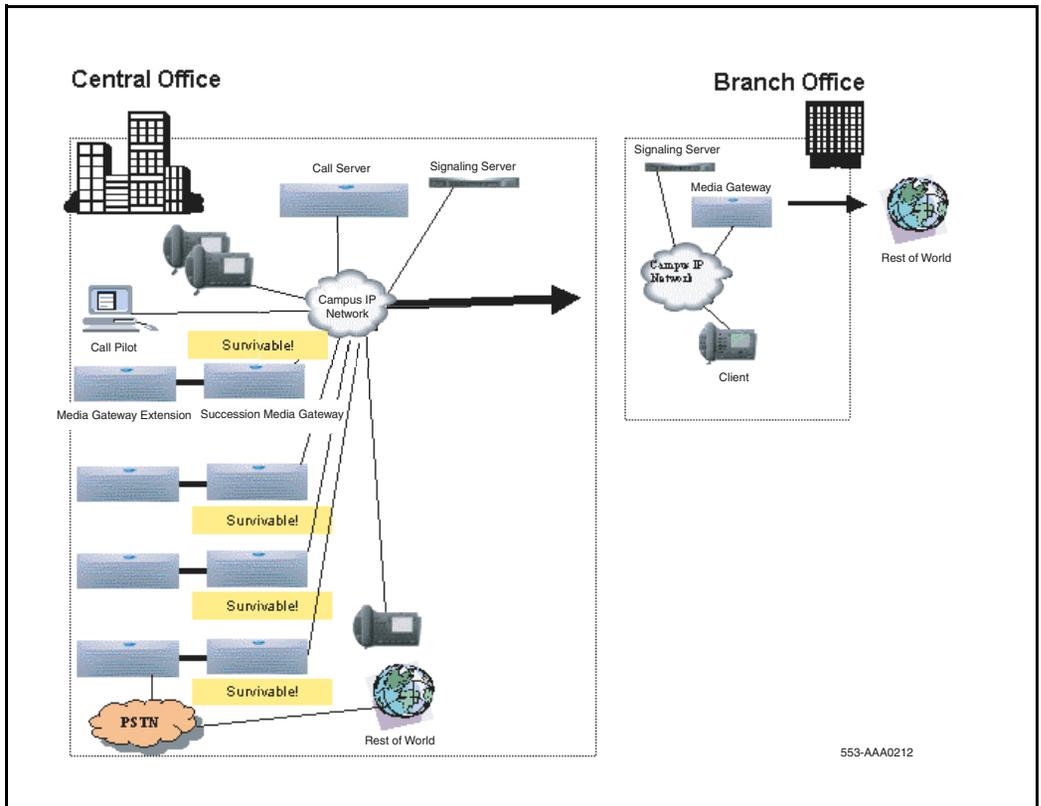
The multi-site deployment with centralized call processing consists of the following:

- Call Server at the central site.
- Campus distributed Media Gateway.
 - The Media Gateway can be survivable (Alternate Call Server)
 - Optional Signaling Server.
 - Local trunks PSTN and 911 access.
 - Internet Telephones are configured and managed centrally from the central site.

Multiple sites

Figure 5 shows a multi-site deployment with a central Call Server and a Branch Office.

Figure 5
Central Call Server with Branch Office



The multi-site deployment consists of the following:

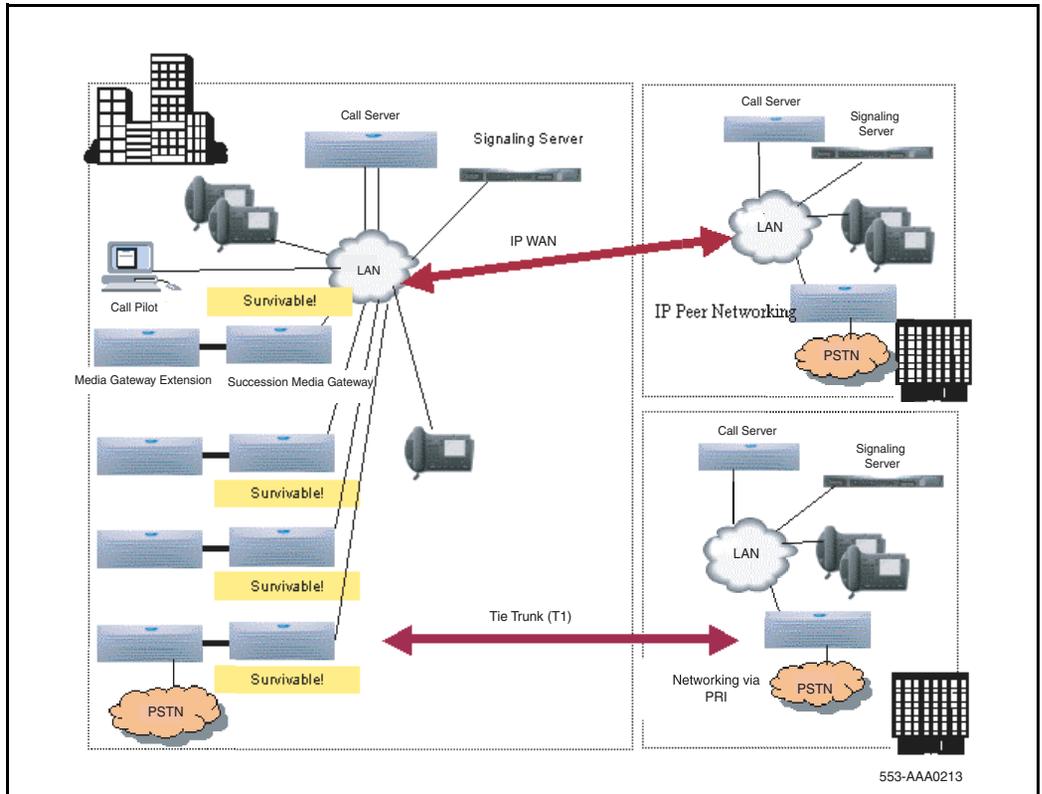
- Call Server at the central site that provides call processing for the main and Branch Offices.
- Succession 1000 Branch Office.
 - Requires its own Signaling Server.

- Media Gateway in remote Branch Offices enables local PSTN and 911 access.
- Internet Telephones are configured and managed centrally from the central site.
- The Media Gateway is survivable in the event of a WAN failure.
- You can configure up to 400 users for survivability.
- You can use the Signaling Server virtual trunking to the central site, for system management purposes.

Distributed Call Servers

Distributed Call Servers are defined as multiple Succession 1000 systems. An example of multi-site deployment with distributed Call Servers is shown in Figure 6 on [page 29](#).

Figure 6
Distributed Call Servers



The multi-site deployment consists of the following peer sites:

- separate Call Servers that provide call processing for each site
- networking supported with IP data network, or circuit-switched (analog or digital) trunks

- full-featured networking between sites using IP Peer Networking and/or tie trunks
- transparent to user at all sites

Capabilities

The Call Server can support up to four Media Gateways, and each Media Gateway can support a Media Gateway Expansion.

The Media Gateway has a Succession System Controller (SSC) card and has four available slots for flexible configurations of line and trunk cards.

The Call Server can support up to 1000 Internet Telephones.

System features

The Succession 1000 supports the full suite of software features. See *Feature Listing* (553-3001-011) for a list of features.

Power over LAN

The Power over LAN uses a Power Inline Patch Panel. The Power Inline Patch Panel has the potential to support the CISCO® set power strategy. The desktop Internet Telephones can obtain their power through the network instead of through their own power transformers. This makes the desktop installation neater, and assists a power failure backup strategy.

Desktop clients and accessories

Internet Telephones

The Succession 1000 supports three models of Internet Telephones. The i2002 Internet Telephones, i2004 Internet Telephones, and i2050 SoftPhones convert voice into data packets for transport using Internet Protocol (IP).

i2002 Internet Telephone

Figure 7 on [page 31](#) shows the i2002 Internet Telephone with an integrated three-port switch. See *Internet Terminals: Description* (553-3001-368) for more information.

Figure 7
i2002 Internet Telephone



i2004 Internet Telephone

Figure 8 shows the i2004 Internet Telephone with an integrated three-port switch. The previous version of the i2004 Internet Telephone requires an external three-port switch. See *Internet Terminals: Description* (553-3001-368) for more information.

Figure 8
i2004 Internet Telephone



i2050 SoftPhone

Figure 9 shows the i2050 SoftPhone. See *Internet Terminals: Description* (553-3001-368) for more information.

Figure 9
i2050 Internet Telephone



Digital telephones

The Succession 1000 system supports a wide range of digital line cards. the line cards provide global telecommunications market interfaces.

The Succession 1000 system supports the following digital telephones:

- M3900 series Meridian Digital Telephones consist of the following:
 - M3901 Entry Level Telephone
 - M3902 Basic Telephone
 - M3903 Enhanced Telephone
 - M3904 Professional Telephone
 - M3905 Call Center Telephone
- The Succession 1000 system conforms to the Evergreen policy and supports the following available digital telephones:
 - M2006

- M2008
- M2008HF
- M2616
- M2016S
- M2216ACD
- M2317 Telephone

For further information, see *Telephones and Consoles: Description* (553-3001-367).

Analog telephones and devices

The Succession 1000 system supports a wide range of analog line cards that provide analog telephones and T.38 Fax interfaces.

IP Adapter

The Succession 1000 system supports the Meridian Internet Telephone adapter package, for the M26xx and M39xx telephones. The Internet Telephone adapter package is intended for local deployment and does not support analog PSTN fallback.

Attendant consoles

The Succession 1000 system supports an Attendant PC software console and the M2250 attendant console.

The Attendant PC software enables users to perform attendant console and call processing functions on a Windows® PC using a mouse or keyboard.

The Attendant PC combines the call-processing power of the M2250 Attendant Console with the information processing and storage power of a PC enhancing attendant services.

For further information, see *Telephones and Consoles: Description* (553-3001-367).

Applications

The following applications work with Succession 1000:

- CallPilot
- Symposium Call Center Server (SCCS)
- SECC
- Symposium TAPI Server
- Meridian Integrated Recorded Announcement (MIRAN)
- Symposium Agent
- Symposium Web Center Portal
- Remote Office
- Wireless
 - e-Mobility 802.11 wireless LAN
 - Meridian Digital Enhanced Cordless Telecommunications (MDECT)
- Meridian Integrated Call Assistant (MICA)
- Meridian Integrated Conference Bridge (MICBII and MICBIII)
- Meridian Integrated Personal Call Director (MIPCD)
- Meridian Integrated Voice Services (MIVS)

CallPilot applications

CallPilot is a multimedia messaging system that offers a single solution for managing information, including the following:

- voice, fax, and e-mail messages
- telephone calls (including conference)
- calendars
- directories
- call logs

CallPilot 2.0

CallPilot 2.0 includes the 501t and the 1002rp messaging platforms as replacements for the existing standalone platforms.

CallPilot 1.5 Mini

CallPilot 1.5 Mini is a less expensive, embedded messaging stand-alone platform that provides Unified Messaging and base Voice Messaging capabilities in both Meridian 1 and Succession 1000 switching environments. For further information, see *CallPilot (555-7101-201)*.

Symposium Call Center Server

Symposium Call Center Server (SCCS) offers a suite of applications that includes call processing and agent handling, extensive management and reporting capabilities, third-party application interfaces, and real-time displays for supervisors and managers. For further information, see the Symposium Call Server documentation.

Meridian Integrated RAN (MIRAN)

MIRAN enables the user to manage recorded announcements using a Browser User Interface (BUI), a Telephone User Interface (TUI), or a text-based user interface. For further information, see *Meridian Integrated RAN: Description, Installation, and Operation (553-3001-360)*.

Remote Office 9150

Remote Office enables remote employees of central offices to access Succession 1000 features and functionality using an IP WAN. The Remote Office unit installs at the remote site and communicates with the central site using a 10BaseT Ethernet or ISDN Basic Rate Interface (BRI) connection. It uses VoIP technology to route voice and signaling packets between the remote office site and the Succession 1000 main site, enabling seamless integration.

The VoIP features automatically switch from the IP network to the PSTN, when voice Quality of Service (QoS) falls below a pre-determined threshold. IP Telephony resumes when QoS levels return to an acceptable level.

Remote Office 9150 supports a maximum of 32 digital telephones.

For further information, see *Remote Office 9150 Installation and Administration Guide* (555-8421-215).

Meridian Integrated Call Assistant

The Meridian Integrated Call Assistant (MICA) can route calls to desired destinations after a series of greetings are played. The destination is based on the caller's telephone keypad strokes. For further information, see *Meridian Integrated Call Assistant: Engineering, Installation, Administration, and Maintenance* (553-3001-362).

Meridian Integrated Conference Bridge

The Meridian Integrated Conference Bridge (MICB) enables a user to schedule and manage conference bridges using a Browser User Interface (BUI) or a Telephone User Interface (TUI). These interfaces enable a user to schedule and configure one-time and recurrent conferences. For further information, see *Meridian Integrated Conference Bridge: Service Implementation Guide* (553-3001-358).

Meridian Integrated Personal Call Director

The Meridian Integrated Personal Call Director (MIPCD) enables users to automatically forward incoming telephone calls to another number, such as a cellular or home telephone. The MIPCD continues to forward the call until the call is answered, or all of the options are exhausted. For further information, see *Meridian Integrated Personal Call Director: Description, Installation, Administration, and Maintenance* (553-3001-361).

Meridian Integrated Voice Services

Meridian Integrated Voice Services (MIVS) provides the hospitality services of Automatic Wake Up (AWU) and Do Not Disturb (DND). The guest dials an access DN and follows instructions until a confirmation message plays to activate a feature. For further information, see *Meridian Integrated Voice Services: Description, Installation, Administration, and Maintenance* (553-3001-359).

Interworking/Interoperability

The Succession 1000 uses the standard H.323 protocol to support interworking with other vendors. Interoperability testing is required to determine which products are compatible with Succession 1000 systems. There are a number of products that are supported as part of the development, including the following:

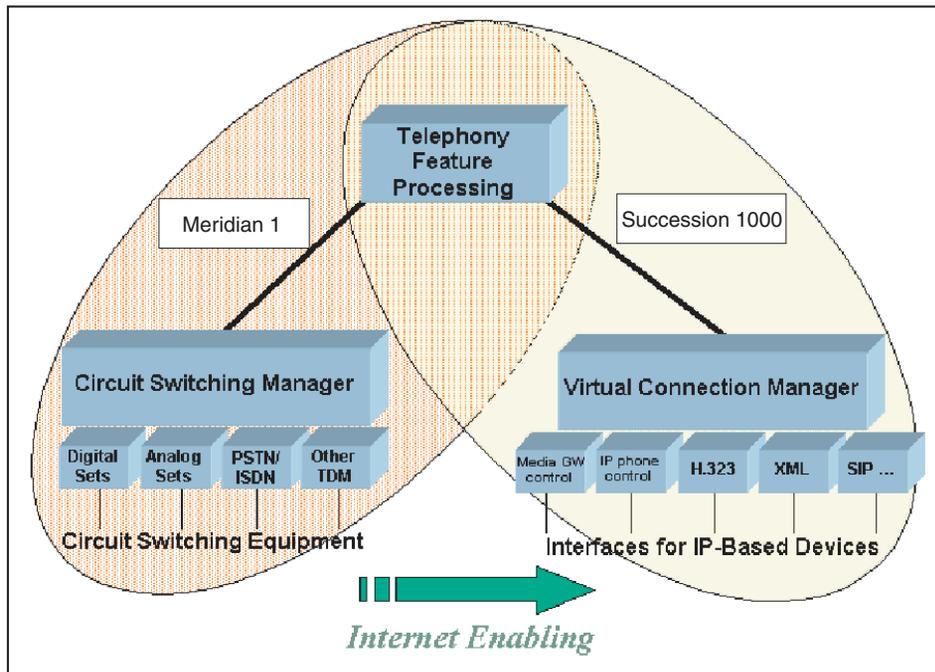
- BCM 2.5+
- Succession 1000M using an IP Trunk 2.x up-issue

Interworking with Nortel Networks' Succession 1000M and BCM products is provided to enable enterprise networks to smoothly migrate from traditional TDM-based PBXs to IP-based telephony products.

Migrating to an IP telephony network

Succession 1000 architecture provides an IP-based alternative to the Meridian 1 architecture. It is designed to enable IP-based communications models in the future, while protecting the investment of Meridian 1 end-users as they migrate to new technologies. Figure 10 on [page 39](#) illustrates the key components of this architecture.

Figure 10
Software architecture for Telephony



Software Architecture for Telephony

Telephony Feature Processing is based upon the existing Meridian 1 Call Processing software and provides the industry-leading feature capabilities that have been built over the years.

The Virtual Connection Manager enables IP-based devices that are registered with the Call Server access to the same feature capabilities as existing telephony devices. In this way, the feature set available on the Meridian 1 system today is made available to evolving IP devices and soft clients, and is the switching foundation of Succession 1000. In addition to access to the Telephony Feature Processing of a Call Server, devices can also access other services in the network, such as XML and IM. The Virtual Connection Manager supports Internet Telephones, and a wider suite of devices and services.

The Circuit Switching Manager evolved from Meridian 1 software. This migration strategy enables Telephony Feature Processing for future releases of Succession 1000 to control existing Meridian 1 hardware components. In this way, the investment in existing equipment is preserved while new capabilities are added at a managed pace.

Planning for system deployment

Contents

This section contains information on the following topics:

Planning for Succession 1000 deployment	41
Succession 1000 numbering plans and call routing	45

Planning for Succession 1000 deployment

In order to ensure that the Succession 1000 system is properly deployed, it is important to first evaluate the existing telecom and data infrastructure.

The telephony features offered in the Succession 1000 are based on the Meridian 1 capabilities. The infrastructure of Succession 1000, however, is not the same as Meridian 1 infrastructure. The performance of the telephony features are only as good as the data network-to-system connections. A technical understanding of data networking and VoIP is essential for optimal performance of the Succession 1000. See *Data Networking for Voice over IP* (553-3001-160)

Evaluate existing telephony infrastructure

Planning for Succession 1000 deployment begins with the evaluation of the existing Telecom infrastructure. See *Succession 1000 System: Planning and Engineering* (553-3031-120).

You can replace existing network components, or add a Succession 1000 system, migrate existing users, and/or add new users.

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

- 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

Evaluate existing data infrastructure

Evaluate existing data infrastructure for suitability when deploying a VoIP solution.

It might be necessary to provide the following infrastructure required for the converging environment:

- additional bandwidth
- consistent performance
- greater availability

Evaluate Both LAN and WAN infrastructures.

To evaluate voice performance requirements, review the following:

- device inventory
- network design
- baseline information

Links and devices must have sufficient capacity for additional voice traffic. Upgrade links with high peak or busy hour utilization.

Evaluate devices with the following characteristics:

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

Peak utilization characteristics in the baseline can reveal potential voice quality issues.

To evaluate availability requirements for the VoIP network, review the following:

- network topology
- feature capabilities
- protocol implementations

Review redundancy capabilities of the network to ensure that availability goals meet the current network design (or a new design) recommended for IP Telephony.

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

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

- memory
- bandwidth
- features

Knowledge requirements

Areas of knowledge needed to plan and deploy a Succession 1000 system include the following:

- System capabilities
 - feature operation
 - configuration
 - numbering/dial plan configuration
 - applications (such as Call Pilot and Symposium)
- VoIP technology
 - H.323 protocols
 - VoIP concepts and protocols
 - RTP
 - Codecs (such as G.711 and G.729)
- Data network architecture
 - TCP/IP
 - IP subnetting
 - routing protocols (such as EIGRP, OSPF, RIP, and BGP)
- Data services/peripherals
 - DNS
 - DHCP
 - TFTP
 - Web server
 - QoS
- Access
 - signaling (ISDN-PRI, EIR2, CCS, and CAS),

- FXS
- FXO
- ground/loop start
- Recommended experience
 - VoIP configuration
 - Ethernet switches
 - QoS techniques
 - knowledge and understanding of voice dial plans
 - exposure to Meridian 1

Succession 1000 numbering plans and call routing

There are two primary mechanisms for call routing, as follows:

- the PBX numbering plan
- the Gatekeeper for routing

PBX numbering plan and routing

When a user on a Succession 1000 system 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 routes the call in one of two ways:

- The system uses UDP or CDP to route the call to the proper trunk group.
- The system uses Vacant Number Routing (VNR) to route the call to a Gatekeeper. With VNR, any number not in the system numbering plan is routed to a specific trunk route, and a Gatekeeper centralized database determines the destination of the call.

Gatekeeper routing

Once it is determined that a call is sent over an IP network, the call is routed to the H.323 gateway software, which uses the Gatekeeper to route the call. The Gatekeeper translates the address from a telephone number to an IP signaling address, and authorizes the call in a H.323 network.

Zoning plan

In an H.323 network, each Gatekeeper controls one zone. Each H.323 zone can consist of many H.323 IP clients (Internet Telephones), and many Voice Gateway Media Cards. If a call terminates beyond the call's own zone, the Gatekeeper of another zone must set up the connection.

It is possible to divide a Succession 1000 system into several zones. It is possible to divide a customer within a system into different zones. However, it is more common is to assign one zone to one system and one customer.

Numbering plan options

You can, when planning a Succession 1000 network, chose many numbering plans, depending upon customer dialing preferences and configuration management requirements.

Uniform Dialing Plan

In a Uniform Dialing Plan (UDP), each network location is assigned a Location Code (LOC). Each telephone has a Directory Number (DN) that is unique within its Call Server (and customer). To reach a user, you must know the LOC and 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, a user would dial: 6 343 2222.

With a UDP numbering plan, the Gatekeeper must keep the Home Location Code (HLOC) of every Media 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 Media Gateway. The Gatekeeper searches for the Location Code within its database and returns the IP addressing information for the site. Then the Media Gateway software can directly set up a call to the desired Media Gateway.

Gatekeeper role

The Gatekeeper performs the following functions:

- Translates an address from an alias (in this case, a telephone number) to an IP signaling address.
- Authorizes the call in the H.323 network.

Coordinated Dialing Plan

In the Coordinated Dialing Plan (CDP), each location is allocated one or more “Steering Codes” that are unique within a CDP domain. This enables a user’s DN on a number of Call Servers to be reached with a fairly short dialing sequence. The DN of each user (including the steering code) must be unique within the CDP domain.

For example, Call Servers can be coordinated to support five-digit dialing within a campus environment. In this example, the steering code allocation could be as follows:

- Call Server A:Steering codes 3 and 4 (for example, DN’s in the range 3xxxx and 4xxxx)
- Call Server B:Steering code 5 (for example, DN’s in the range 5xxxx)

Within this group of Call Servers, users can dial their unique DN to call each other. However, all DNs on Call Server A must be in the range 3xxxx or 4xxxx, whereas all DNs on Call Server B must be in the range 5xxxx. If a user moves from one Call Server to another, their DN must change.

Dialing AC1 to reach UDP numbers and dialing CDP DNs when dialing within a CDP domain CDP can be used in conjunction with UDP.

Transferable Directory Numbers (TNDN)

In a Transferable Directory Numbers (TNDN) dialing plan, each individual user is given a unique DN, which does not change if they move to a different Call Server. The Gatekeeper keeps track of each TNDN in the network so that it knows which Media Gateway(s) to return to when asked to resolve a TNDN address.

With a TNDN numbering plan, Succession 1000 networks enable users to move from location to location while retaining their Directory Number.

Off-Net Call Routing operation

When dialing calls to PSTN interfaces, the Call Server determines that the call is destined Off-Net, based upon digit analysis that must be configured at major Call Servers in the network. This enables the Media Gateway software to request the location of public E.164 numbers from the Gatekeeper. The Gatekeeper is configured with a list of potential “alternate routes” that can be used to reach a certain telephone number. Each route is configured with a “Gatekeeper Cost Factor” to help it determine the least-cost route.

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

When a Gatekeeper replies to a Media Gateway with the address information for E.164 numbers, it provides a list of alternative Media Gateways, sorted in order of cost. If a Media Gateway is busy when a call attempt is made, the originating Media Gateway tries the next alternative in the list. If no alternative is available over the IP network, the originating Call Server can step to the next member on its route list, which could be a PSTN or TIE alternative route.

For example, if an IP network outage occurs that does not enable voice calls to terminate over the IP network, calls would be rerouted to alternate PSTN or TIE routes.

Routing to and from local Branch Office Media Gateways

Internet Telephone users can be located at a Branch Office that is equipped with a Branch Office Media Gateway. Call routing to the local Media Gateway is important, especially if toll charges are applicable to calls from the central Call Server controlling the Internet Telephone. The administrator can configure digit manipulation for Internet Telephones that are located near a Branch Office Media Gateway, in order to select a Media Gateway that provides local PSTN access.

Calls from the PSTN to users within the network can be routed using either (a) the various ESN numbering plan configurations, or (b) a new feature called Network Number Resolution, based upon the Vacant Number Routing feature. This enables small sites, such as those using the Branch Office Media Gateway, to require minimal configuration to route calls through other Call Servers or the Gatekeeper. See *Branch Office* (553-3001-214).

System component description

Contents

This section contains information on the following topics:

Succession 1000 system LAN connections	52
Succession 1000 system components	53
NTDU08 Call Server	55
NTDU27 Signaling Server	64
Signaling Server software applications	67
NTDU14 Media Gateway	73
NTDU15 Media Gateway Expansion chassis	80
Ethernet switch (customer-supplied)	83
Power over LAN (optional)	83
Internet Telephones	86
NTVQ01 Succession Media Card	88
Software architecture	92
System management	94
Element Manager	94
IP Line 3.0 application	91
Call Server and Media Gateway software	92
Signaling software	92
Voice Gateway Media Card loadware	92
i2002 and i2004 Internet Telephone firmware	93
i2050 Internet Telephone application	93
Software delivery	93
Centralized Automatic Software Upgrade	93
Centralized upgrade	94

Centralized patching	94
File uploading	94
Patching implementation	94
Call Server management	95
Media Gateway management.	96
Signaling Server management	96
Voice Gateway Media Card management	97
Call Server configuration	97

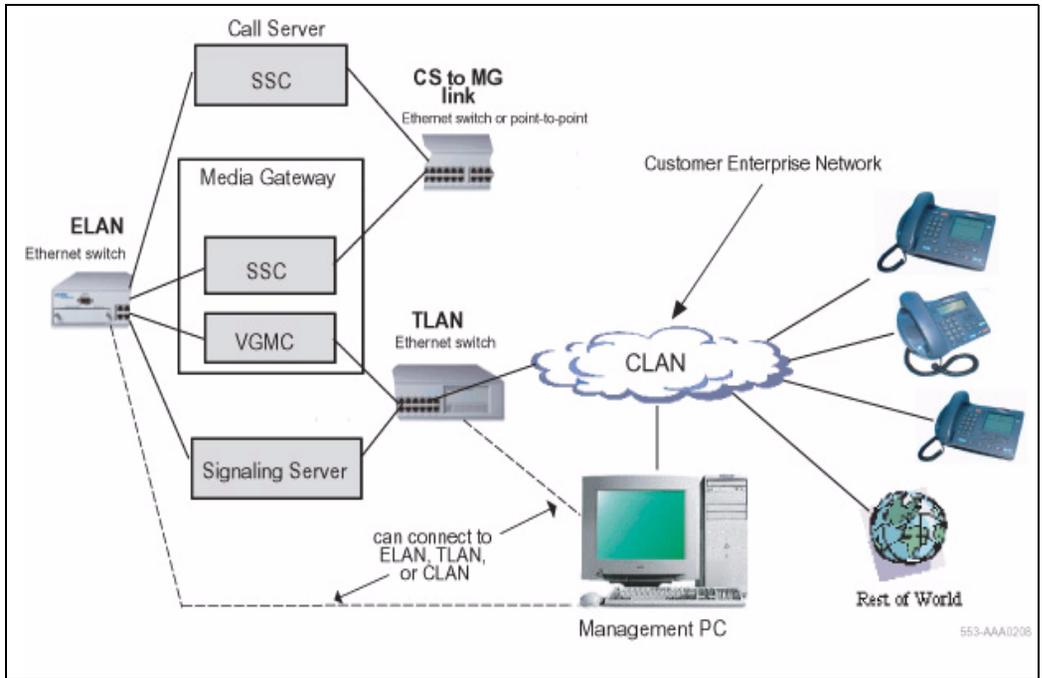
Succession 1000 system LAN connections

Figure 11 on page 53 shows a schematic representation of the Succession 1000 component connections to the various LANs, as follows:

- Embedded Local Area Network (ELAN). The ELAN is a 10BaseT full-duplex LAN, used for management and signaling traffic between the various Succession 1000 components.
- Call Server to Media Gateway Link (CS to MG IP Link). You can connect the CS to MG IP Link point-to-point or through a switch. The CS to MG IP Link carries signaling and telephony traffic.
- Telephony Local Area Network (TLAN). The TLAN is a 100BaseT full duplex LAN, used for telephony voice and telephony signaling traffic. The TLAN is separate from the CLAN. It connects the Signaling Server and VGMCs to the CLAN, and isolates the IP Telephones node interface from CLAN broadcast traffic.
- Customer LAN (CLAN). The CLAN is the customer's IP data network. The Internet Telephones are connected to the CLAN. Access to the WAN is provided through the CLAN. You can extend the CLAN to other customer sites over WAN links.

See *Data Networking for Voice over IP* (553-3001-160) for more details.

Figure 11
Succession 1000 connections



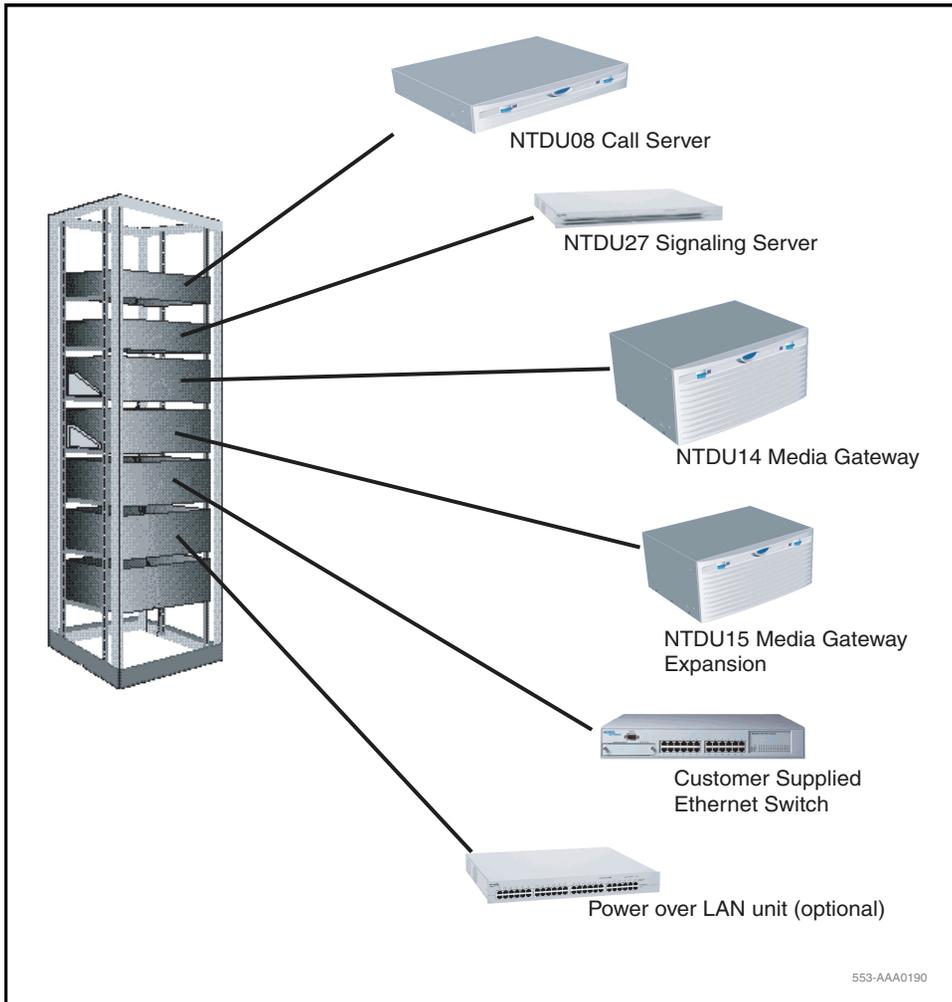
Succession 1000 system components

A typical system consists of the following major hardware components:

- NTDU08 Call Server
- NTDU27 Signaling Server
- NTDU14 Media Gateway
- NTDU15 Media Gateway Expansion
- i2002, i2004, i2050 Internet Telephones
- customer-supplied Ethernet Layer 2 switch
- Power over LAN unit (optional)

Figure 12 shows the major hardware components installed in a customer-supplied 19-inch rack.

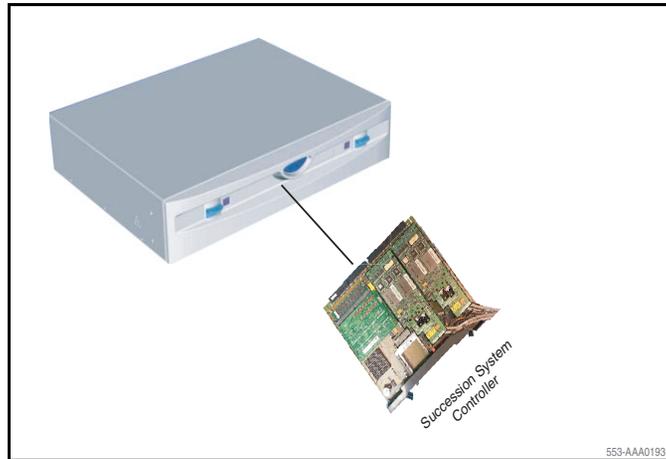
Figure 12
Succession 1000 components in a rack mount



NTDU08 Call Server

Figure 13 shows the Call Server chassis, with a Succession System Controller (SSC) card.

Figure 13
Call Server



A Call Server can control a maximum of four Media Gateways with their Media Gateway Expansions.

A Succession 1000 system with one Call Server supports up to 1000 Internet Telephones. You can network a Succession 1000 system with other Succession 1000 systems to support larger numbers of Internet Telephones.

Figure 14
NTDU08 Call Server front features

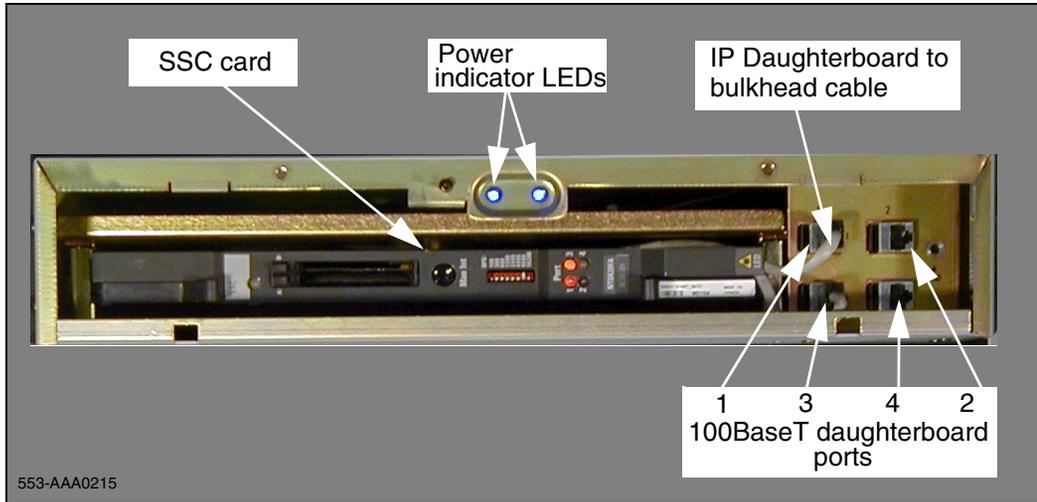


Figure 14 shows the Call Server chassis with the front cover removed. The front features are as follows:

- The NTDK20 SSC is the only card in the Call Server.
- The power indicator LEDs light when the power to the Call Server is on. The LEDs show through the front cover logo.
- The IP daughterboard bulkhead cables connect one or two 100BaseT IP daughterboards to the Call Server bulkhead ports. The IP daughterboards are located on the component side of the SSC card, as shown in Figure 18 on [page 63](#). Each IP daughterboard has two ports.

Figure 15
NTDU08 Call Server rear connectors

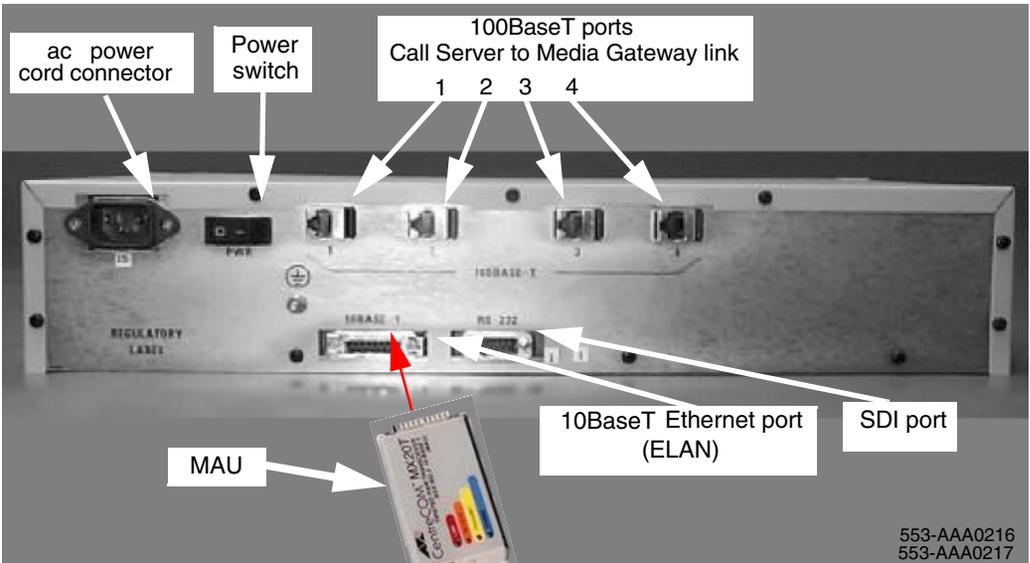


Figure 15 shows the connectors located on the rear of the Call Server. The function of the connectors are as follows:

- The ac power cord connector provides an ac connection to the Call Server.
- The power switch controls the ac power to the Call Server.
- The Call Server to Media Gateway Link provide connections for up to four Media Gateways.
- The ELAN port requires a Medium Access Unit (MAU) to connect the Call Server to the ELAN.
- The SDI port provides a 3-port SDI connector for modems, maintenance terminals (TTYs), CDRs, and other SDI devices.

NTDK20 Succession System Controller card

The Succession System Controller (SSC) card in the Call Server is the primary system processor. It controls the telephony services and call processing. Each Media Gateway is also equipped with an SSC card.

The Call Server SSC controls the call processing features of Internet Telephones and trunk interfaces when in normal mode of operation.

The Call Server SSC synchronizes the configuration information on all Media Gateway SSC cards. The configuration data on all Media Gateway SSC cards always exactly matches the Call Server SSC configuration data.

All Media Gateway SSC cards are also synchronized with the call processing on the Call Server SSC card. This synchronization enables a Media Gateway SSC card take over local call processing if the primary Call Server fails to respond.

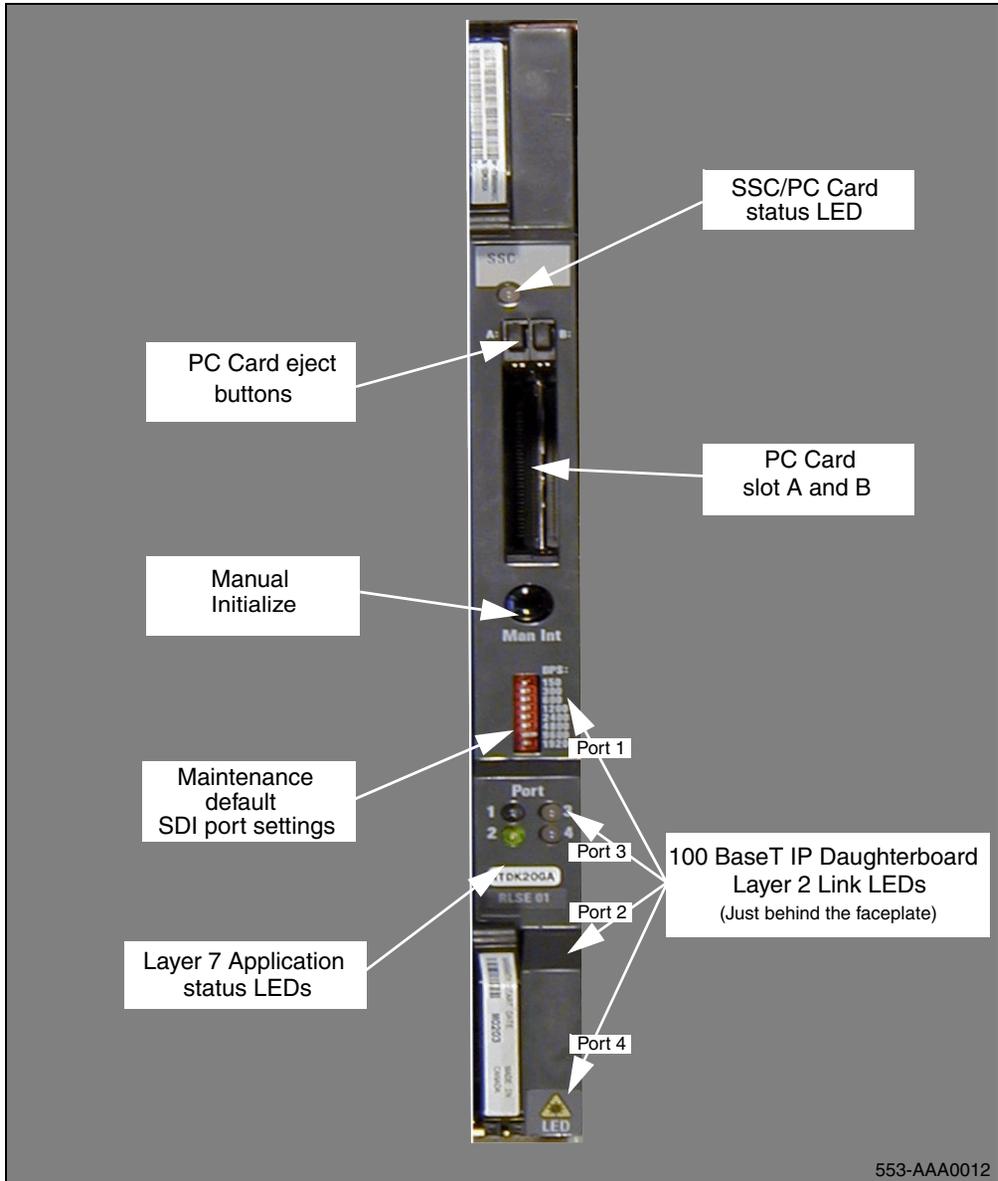
Figure 16 on [page 59](#) shows the SSC card faceplate features, as follows:

- The SSC/PC Card LED indicates status.
- The PC Card eject buttons enable the removal of the PC Cards.
- The PC Card slots provide access for one Type III or two Type II PC Cards.

Note: Nortel Networks supply PC Card adaptors which enable you to use Compact Flash cards to in this slot for software delivery.

- The Manual Initialize button provides a call processing software warm start.
- The IP 100BaseT daughterboard port LEDs indicate status.

Figure 16
SSC card faceplate



Faceplate LEDs

The NTDK20 SSC card has either three or five faceplate LEDs, depending on the version of the card.

The SSC/PC card LED indicates the following, if the LED is:

- Off, the SSC is in normal operation
- Yellow, the SSC is disabled
- Red, the SSC is running self-test
- Red and flashes three times, the self-test passed
- Green steady or flashing, the PC card is accessed

The Layer 7 Port LEDs indicate the following, if the Port LEDs are:

- Red, the link is disabled and voice is disabled
- Amber, the link is established and voice is disabled
- Green, the link and voice are established

The IP Daughterboard port Layer 2 Link LEDs indicate the following:

- Green, the link is established
- Red (receive) and yellow (transmit) flashing, show network activity

Figure 17
SSC card IP daughterboard LEDs

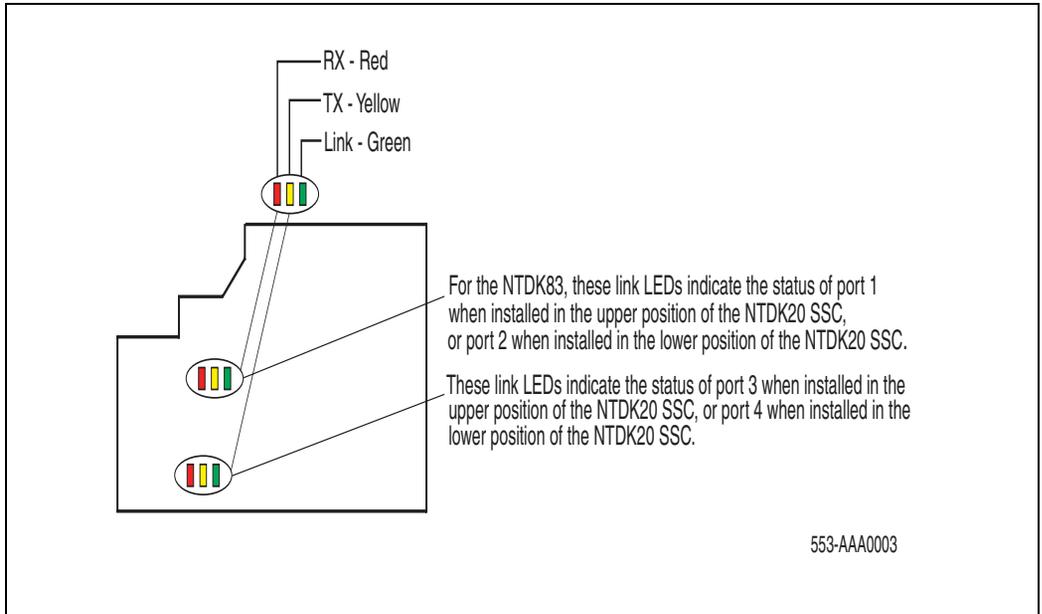
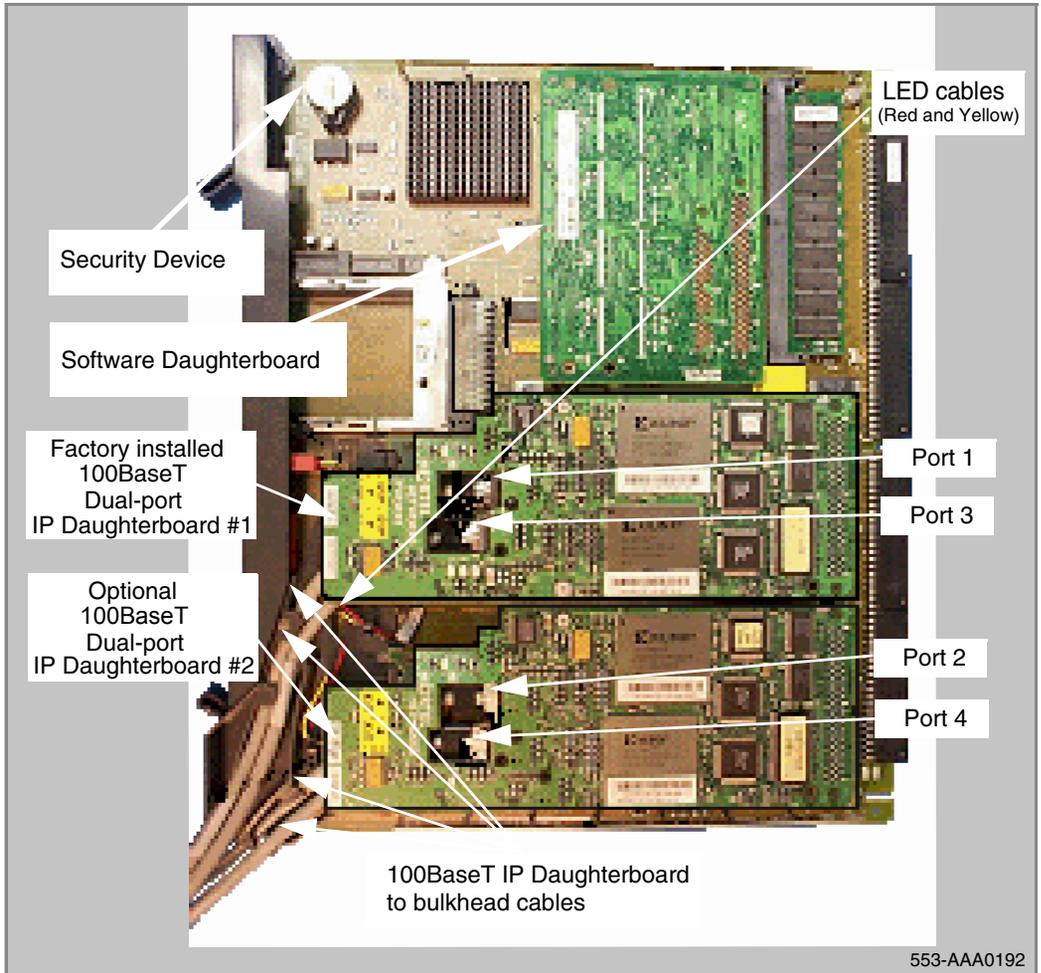


Figure 18 on page 63 shows the Call Server SSC card components. The function of the components are as follows:

- CPU (not shown) is comprised of two processors. The main processor handles call processing, serial ports, and network traffic.
- Auxiliary processor (not shown) handles card polling, power monitoring, tone generation, and control of a Digital Signal Processor (DSP) for tone detection.
- Ethernet controller (not shown) provides one port between the CPU and ELAN.
- Tone and Digit Switch (not shown) provides 30 channels of tone generation for the system.
- Digitone Receiver (not shown) provides eight DTR/XTD units with an additional user selectable eight DTR/XTD units, or four MFC, MFE, MFK5, MFK6, or MFR units.

- Security Device enables the activation of features assigned to the system, through the use of a series of keycodes.
- Software Daughterboard stores the system software.
- One factory installed Dual-port IP daughterboard (#1) providing 32 conference channels and an optional Dual-port IP daughterboard (#2) to provide an additional 32 conference channels.
- The 100BaseT bulkhead cables connect the Dual-port IP daughterboard ports 1, 3, and 2, 4 to the Call Server chassis bulkhead connectors.
- The red, yellow, and black LED cables connect the Dual-port IP daughterboards to the faceplate Port LEDs.

Figure 18
Call Server NTDK20 SSC card components



NTDU27 Signaling Server

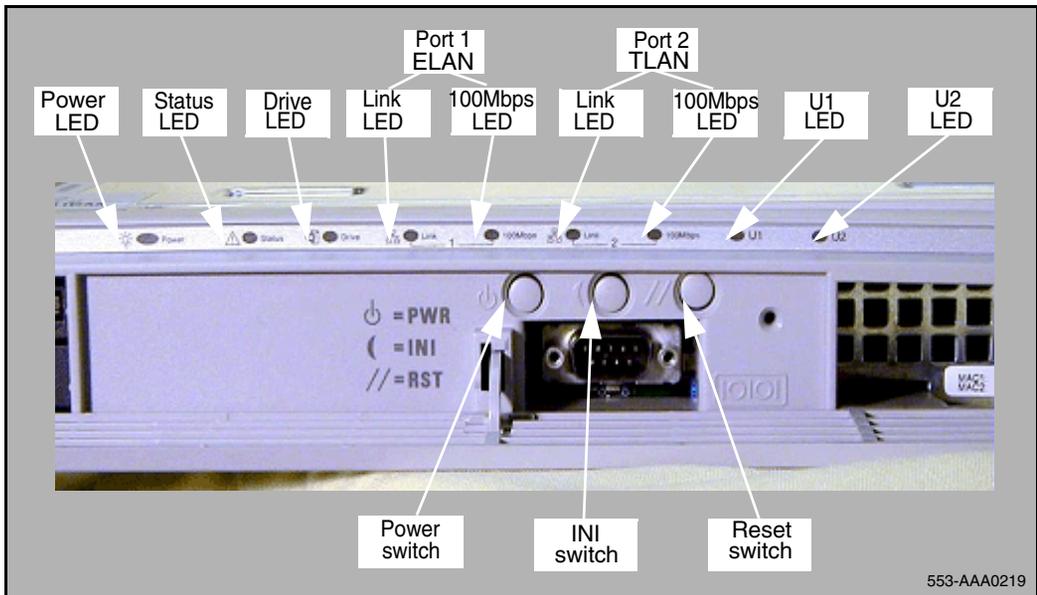
Figure 19 shows the NTDU27 Signaling Server chassis. There are no user serviceable components in the Signaling Server. Opening the Signaling Server voids the warranty.

Figure 19
Signaling Server



Figure 20 and Figure 21 on page 66 show the Signaling Server front features.

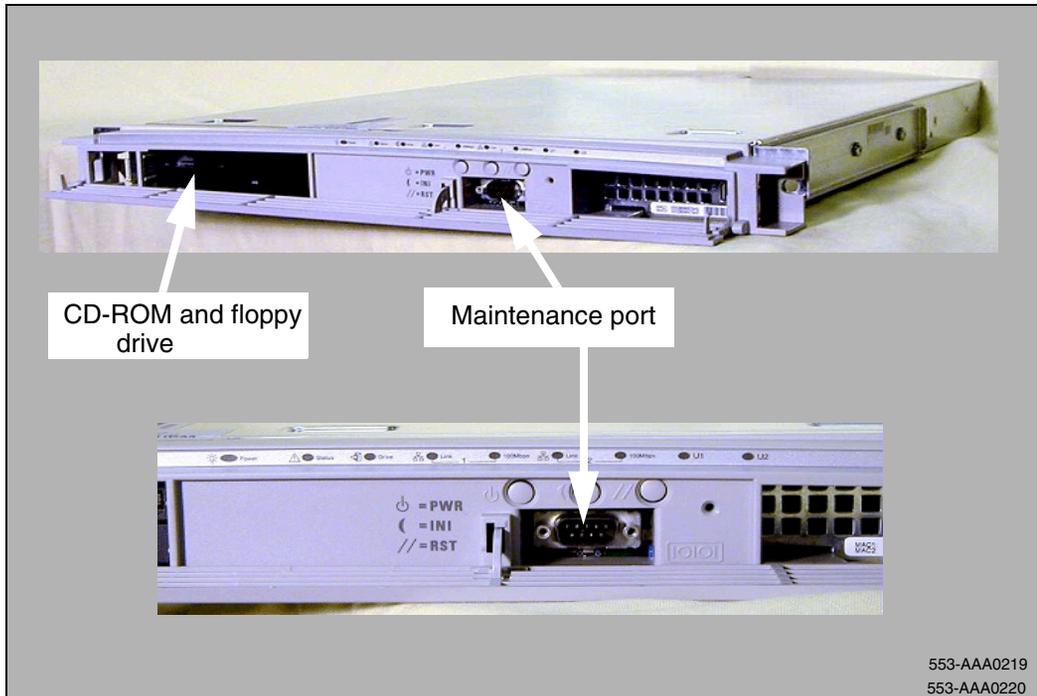
Figure 20
Signaling Server LED indicators and power switch



The Signaling Server front features, as seen in Figure 20 on page 64, show the following:

- Power LED green - ac power on. Power LED off - ac power off.
- Status LED red - CPU stopped. Status LED off - CPU running.
- Drive LED flashing green - Hard Drive or CD ROM Drive active. Drive LED off - Hard Drive or CD ROM Drive inactive.
- Link LED green - Ethernet port active. Link LED off - Ethernet port inactive.
- 100Mbps green LED - Ethernet port running at 100Mbps. 100Mbps LED off - Ethernet port running at 10Mbps.
- The power switch controls the ac power to the Signaling Server.
- The Reset switch invokes a cold reboot of the Signaling Server.
- The INI switch is reserved for future use.

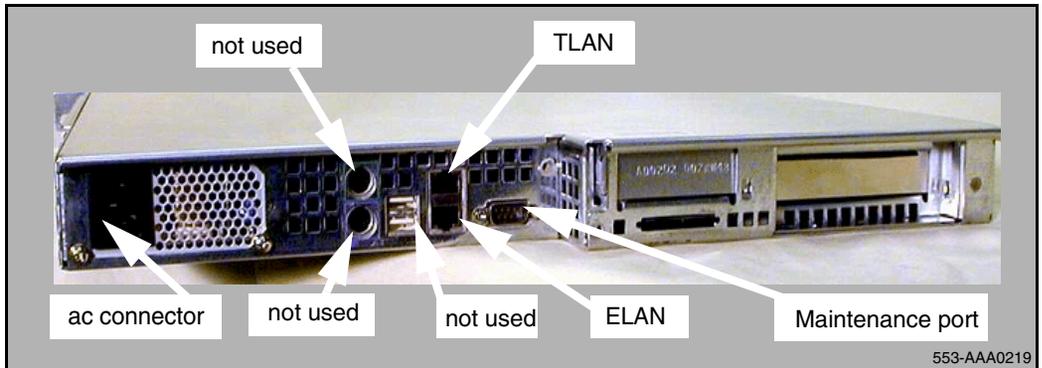
Figure 21
Connectors on the front of the Signaling Server



The Signaling Server front features, shown in Figure 21, are as follows:

- The CD-ROM drive is used to load the Signaling Server Software software files for the Signaling Server, VGMCs, and Internet Telephones. The Signaling Server software includes the Signaling Server operating system, and applications, and all Element Management web server files.
- The floppy drive is used if the CD-ROM is not bootable. To create a boot floppy, use the files in the `mkboot` directory on the Signaling Server Software CD-ROM. You can use the same boot floppy for any or all Software CD-ROMs.
- The front Maintenance port does not display system messages. However, this port provides a login session for Command Line Interface (CLI) management.

Figure 22
Connectors on the back of the Signaling Server



The Signaling Server rear features, seen in Figure 22, are as follows:

- The ac power cord connector provides an ac connection to the Signaling Server.
- The 100BaseT TLAN connector is used for telephony signaling traffic.
- The 10BaseT ELAN connector is used for connection between the various Succession 1000 components.

Note: The Signaling Server TLAN/ELAN ports connect through a Layer 2 Switch to access other Succession 1000 chassis TLAN/ELAN ports.

- The rear maintenance port is the primary port for maintenance and administration terminals.
- The remaining ports are not used for any Succession 1000 function. Do not plug any device into these ports.

Signaling Server software applications

The Signaling Server provides signaling interfaces to the IP network using software components that run on the VxWorks real-time operating system. You can install the Signaling Server in a load-sharing, survivable configuration for higher scalability and reliability.

The Signaling Server is equipped with the following software applications:

- IP Line 3.0 (Terminal Proxy Server)
- H.323 Signaling Gateway (virtual trunk)
- H.323 Gatekeeper
- VxWorks command line interface (various shells)
- Succession 1000 Element Management web server

IP Line 3.0

The IP Line Terminal Proxy Server (TPS) acts as a signaling gateway between the Internet Telephones and a virtual line card's TNs (Note), and provides the following:

- **UNIStim Line TPS (LTPS)**. The LTPS converts the Internet Telephone UNIStim messages (generated when the user goes off-hook, presses keys, goes on-hook), into messages the Call Server can interpret, and provide telephony features to the Internet Telephones.

Note: The complete term UNIStim is Unified Networks Internet protocol Stimulus

- **Connect Server**. Services all initial connection requests from Internet Telephones and balances them by referring them to a specific Signaling Server.
- **SNTP (Simple Network Time Protocol)**. The IP Telephony node Leader (usually the Signaling Server) receives the system time and date from the Call Server. All Followers (VGMCs and additional Signaling Servers in this IP Telephony node) receive the time from their SNTP client contacting the leader SNTP server.
- **BOOTP server**. The Leader sends IP addresses using the BOOTP server to the VGMCs and Follower Signaling Servers. Every node Leader runs a BOOTP server for its node components.
- **TPS Resource Manager (Virtual Terminal Manager)**. The TPS Resource Manager, in a node, decides where the Internet Telephones should register, and provides:
 - Application access to the Internet Telephones.

- Manages all the Internet Telephones between the applications and the stimulus messaging to the Internet Telephone.
- Maintains context sensitive states of the Internet Telephone. For example, display or lamp state.
- Isolates Internet Telephone-specific information from the applications, such as the number of display lines, number of characters for each display line, tone frequency and cadence parameters.

H.323 Signaling Gateway (virtual trunk)

Virtual trunks are software components configured on virtual loops, similar to Internet Telephones. A virtual trunk acts as the bridge between existing call processing features and the IP network. It enables access to all trunk routing and access features that are part of the MCDN networking feature set. Virtual trunks do not require dedicated Digital Signaling Processor (DSP) resources to provide these features. Virtual trunks include all of the features and settings available to ISDN Signaling Link (ISL)-based TIE trunks, and are configured within trunk routes. Voice Gateway Media Card resources (described on page 88) are only allocated for virtual trunks when it is necessary to transcode between IP and circuit-switched devices.

Note 1: The H.323 Signaling Gateway must register with the H.323 Gatekeeper.

Note 2: Virtual line TNs and virtual trunk TNs enable you to configure service data without hard wiring Internet Telephones or virtual trunks to the Succession 1000 system. The virtual TNs are configured in LD 97. The Succession 1000 system can support up to 1000 virtual line TNs and 200 virtual trunk TNs.

H.323 Gatekeeper

All Succession 1000 systems in the IP Peer network must register to an H.323 Gatekeeper. H.323 Gatekeeper registration eliminates the need for manual configuration of IP addresses and numbering plan information at every site. The H.323 Gatekeeper manages a centralized numbering plan for the network, enabling simplified management of the Succession 1000 network. The H.323 Gatekeeper software identifies the IP addresses of PBXs and H.323 Gateways based on the network-wide numbering plan.

Note: Within each Call Server you must configure the Numbering Plan information required for the Call Server software to internally route calls, such as routing information for locally accessible numbers.

The Primary Gatekeeper software is equipped on one Signaling Server in the network. You can divide large networks into network zones. Each network zone requires a Primary Gatekeeper. For Succession 1000 systems to gain network access, they have to register to their Primary Gatekeeper.

You can equip an Alternate Gatekeeper on a second Signaling Server for higher reliability. The Alternate Gatekeeper automatically synchronizes configuration data with the Primary Gatekeeper. An Inter-Gatekeeper protocol is also supported to enable higher scalability and regional deployment of networks. There is only one Alternate Gatekeeper in a network. A network can be so large that a Signaling Server dedicated to Gatekeeper duty is required.

The Gatekeeper (Primary and Alternate) can operate in stand-alone mode, without a Call Server connection.

The Gatekeeper provides the following primary services:

- endpoint and Gateway registration
- call admission control
- address translation and telephone number to IP lookup
- centralized numbering plan administration

The Gatekeeper is H.323 compliant and can provide Gatekeeper features to other H.323-compliant Nortel Networks endpoints. For example, Succession 1000 and IP Trunk 3.0 endpoints. The Gatekeeper also supports endpoints that do not support H.323 Registration, Admission, and Status (RAS) registration with the Gatekeeper, such as ITG Trunk 1.0 and 2.x.

You must configure these endpoints as non-RAS endpoints. You must configure a static IP address for these endpoints, as well as the telephone numbers that the endpoints can terminate.

Note: Systems that do not support H.323 RAS procedures and H.323 Gatekeeper procedures are referred to as non-RAS endpoints.

The Gatekeeper consists of the following major components:

- Gatekeeper Network Protocol Module (GKNPM)
- Database Module (DBM)
- Primary and standby databases
- Gatekeeper Element Manager web server

Gatekeeper Network Protocol Module

The GKNPM interfaces with the H.323 stack and is responsible for sending and receiving all H.323 RAS messaging.

Database Module

The Database Module (DBM) is responsible for the following:

- configuring the Numbering Plan
- reading and updating the primary and standby databases on disk
- resolving all registration and admission requests which the GKNPM passes to the DBM
- backup and restore

The Gatekeeper Numbering Plan configuration is stored in XML format in two databases on the Signaling Servers hard disk. The primary database is used for call processing and the standby database is used for configuration s.

Gatekeeper Element Manager web server

The H.323 Gatekeeper has its own Element Manager web server, called Gatekeeper Element Manager. This web server is a separate web server from the main Succession 1000 Element Manager web server. This web server is integrated into the Gatekeeper software. The Gatekeeper configuration is performed by accessing the Gatekeeper web server on the Signaling Server. Accessing the Gatekeeper web server on the Signaling Server preforms the Gatekeeper configuration.

VxWorks shells

The Wind River VxWorks shells provide access to the operating system for maintenance and debug operations.

Succession 1000 Element Manager web server

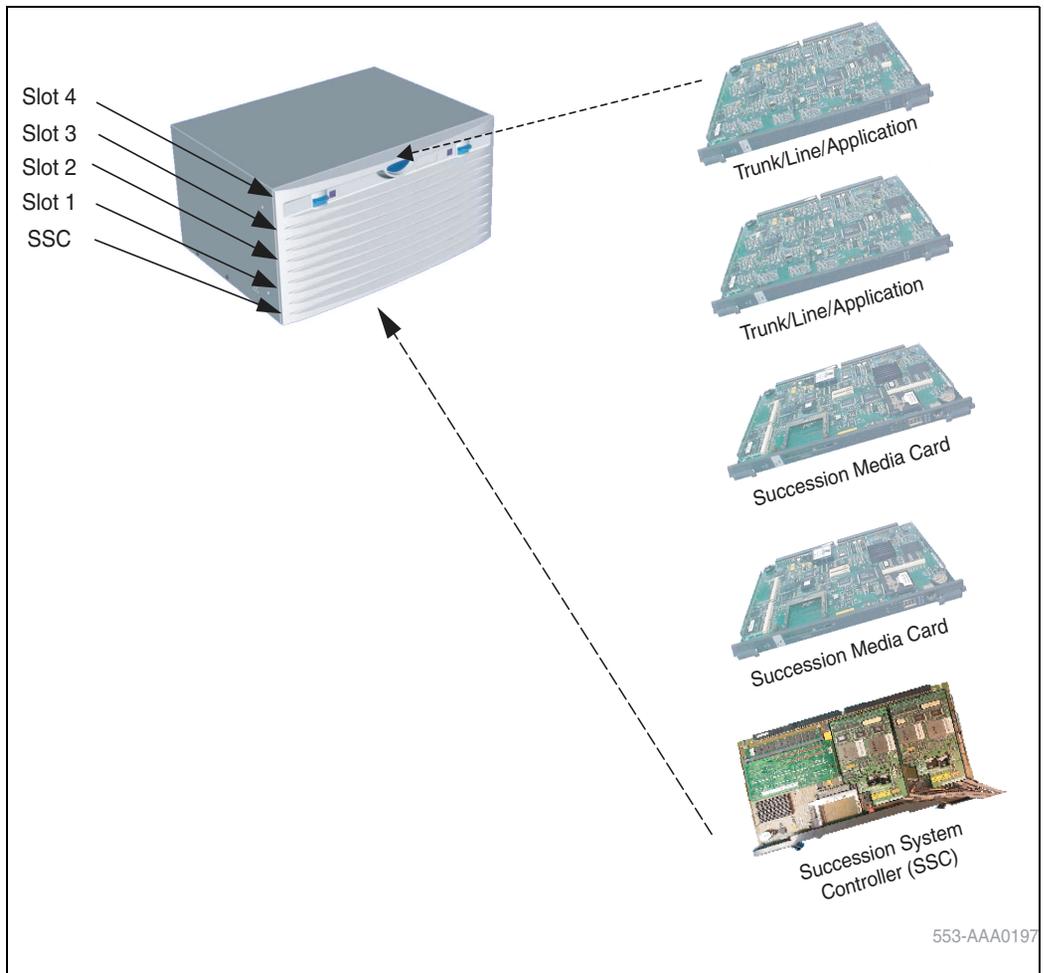
A Succession 1000 Element Manager web server provides a web interface to certain administration and management functions on the Succession 1000 components.

See “Element Manager” on [page 94](#) for more information.

NTDU14 Media Gateway

The Succession Media Gateway supports flexible configurations of PSTN trunks, analog/digital telephone resources, TDM applications cards and Voice Gateway Media Cards. Each Media Gateway can support one Media Gateway Expansion chassis. See Figure 23 on [page 73](#).

Figure 23
Media Gateway



The Media Gateway chassis supports a Succession System Controller card and four slots for flexible configurations of line, trunk, and application cards. Table 1 lists the Media Gateway and Media Gateway Expansion supported cards.

Table 1
Media Gateway and Media Gateway Expansion supported cards

Media Gateway only	Media Gateway/Media Gateway Expansion
NTDK20 SSC (only one SSC card in the bottom slot)	NT5K20 Tone Detector
NTAK02 SDI/DCH	NT5K48 Tone Detector
NTAK03 TDS/DTR	NT8D03 Analog line card
NTAK09 1.5 Mbit DTI/PRI	NT8D09 Message Waiting
NTRB21 1.5 Mbit DTI/PRI	NT8D14 Universal Trunk
NTAK10 2.0 Mbit DTI	NT8D16 Digitone Receiver
NTAK79 2.0 Mbit PRI	NT8D15 E&M Trunk
NTBK50 2.0 Mbit PRI	NTVQ01 Voice Gateway Media Cards
	NT7D16 Data Access
	NT5K02 XFALC
	NT5K18 XFCOT
	NT5K17 XDDI
	NT5K19 XFEM
	NT5K36 XDID/DOD
	NT5K21 XMFC/MFE
	NTAG26 XMFR
	NT8D02 DLC
	Applications cards (application on a NTVQ01 Media Card)

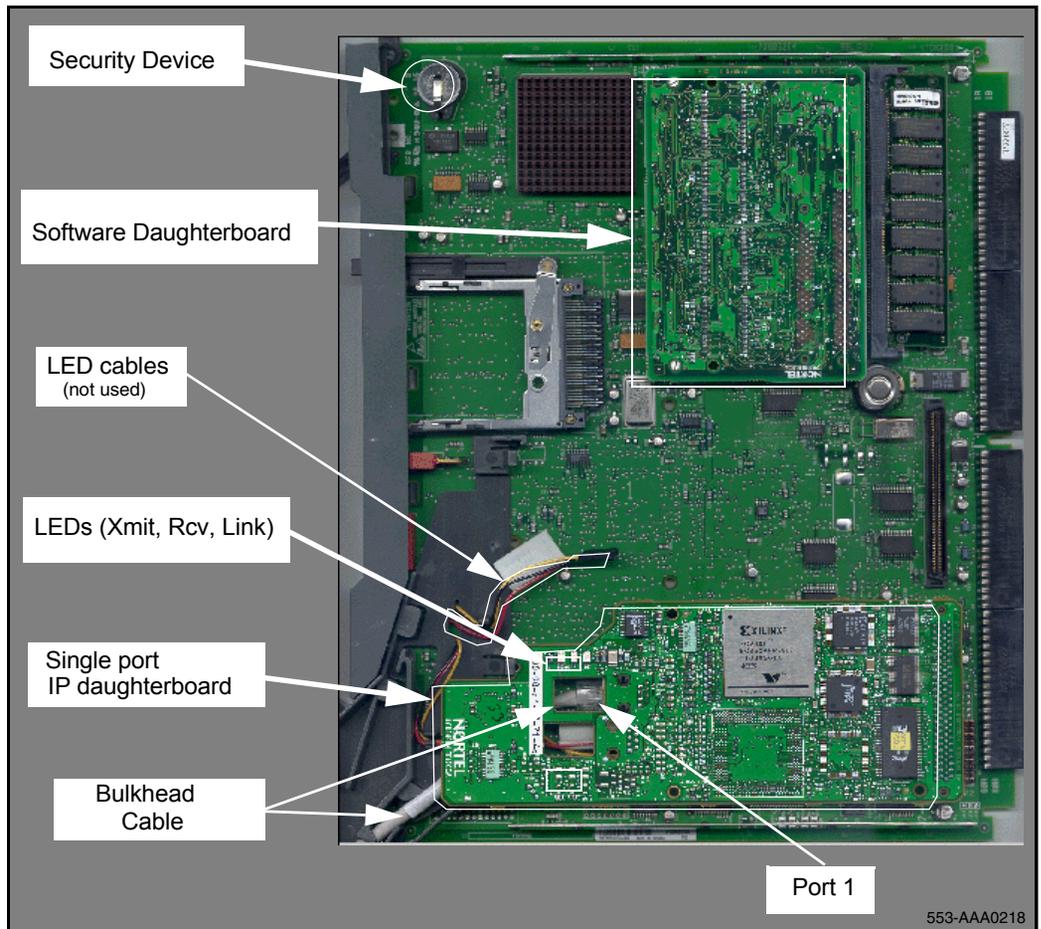
NTDK20 Succession System Controller card

The Media Gateway SSC card has two functions:

- 1 control the interface and application cards
- 2 act as a survivable call processor

The SSC and its components are shown in Figure 24 on [page 75](#).

Figure 24
Location of the Media Gateway NTDK20 SSC card components



The Media Gateway SSC contains software to control the interface and application cards equipped in the gateway while in normal mode of operation. This card also has all hardware resources (Tone, Local Switching and Conference circuits) and software to operate as a survivable call processor if the Call Server fails to respond. The Call Server database is automatically synchronized onto this controller.

If the Call Server fails to respond, each Media Gateway's SSC can become its own independent call processor. This means that in the event of Call Server failure, one Media Gateway SSC does not act as a Call Server for the rest of the system; they are all independent. You must configure a Media Gateway as survivable for this to occur, otherwise it, too, is out of service until the Call Server returns to service. For further information, refer to *Succession 1000 System: Planning and Engineering* (553-3031-120).

Figure 24 on [page 75](#) shows the Media Gateway SSC card components. The function of the components are as follows:

- Security device enables the activation of features assigned to the system, through the use of a series of keycodes. The Media gateway security devices are matched to the Call Server security device.
- Software daughterboard stores the same system software as is on the Call Server.
- One factory installed Single-port IP daughterboard providing 16 conference channels.
- One 100BaseT bulkhead cable connects the Single-port IP daughterboard port 1 to the Media Gateway chassis bulkhead connector 1.
- The red, black, and yellow LED cables are not used on the Media Gateway SSC.

Line and Application cards

Line and Application cards are used to provide a wide range of analog and digital interfaces. A wide range of application platforms are supported from the Meridian 1 product line, such as Call Pilot, MIRAN, and MICB.

Figure 26
NTDU14 Media Gateway connectors

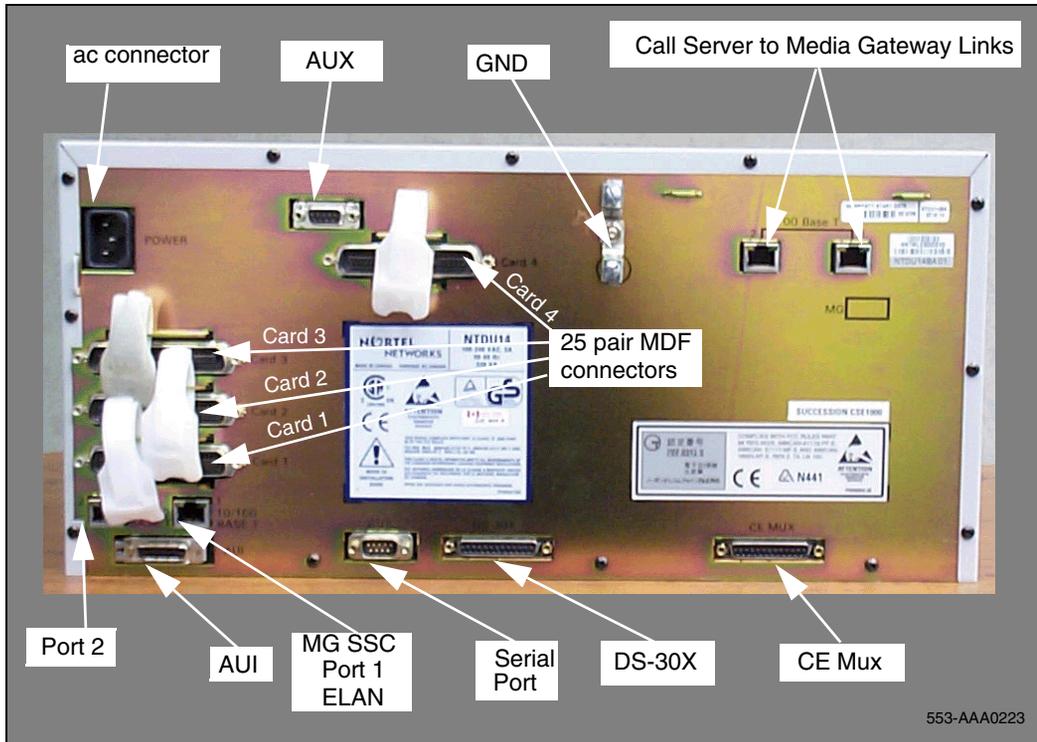
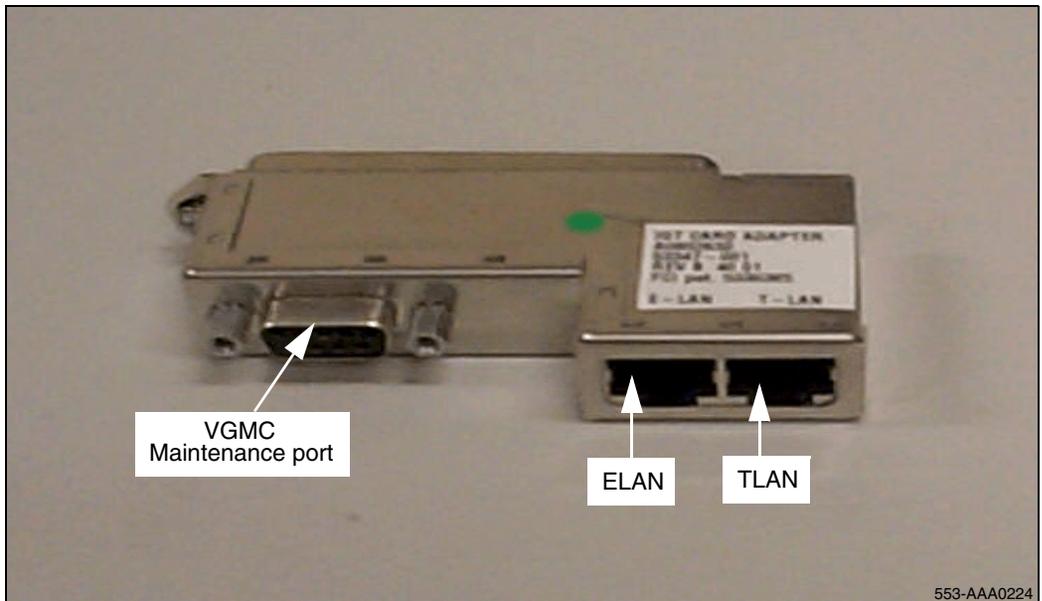


Figure 26 shows the following connectors:

- The ac power cord connector provides ac connection to the Media Gateway.
- AUX is used to extend Power Fail Transfer unit signals to the MDF.
- GND is used for ground cable termination.
- The Call Server to Media Gateway Links provide a connection to the Call Server.
- 10/100BaseT port 2 is reserved for future use.
- AUI (Attachment Unit Interface) is used with earlier version SSC cards which require a MAU.

- 10/100BaseT port 1 connects the Media Gateway SSC to the ELAN.
- SDI port is used to connect maintenance TTYs.
- DS-30X interconnects the Media Gateway and Media Gateway Expansion Peripheral Equipment (card) bus.
- CE Mux interconnects the Media Gateway and Media Gateway Expansion Peripheral Equipment (card) bus.
- 25-pair connectors extend the PE card data to the MDF. The 25-pair connector for the PE card slot where the VGMC is located requires a VGMC Adaptor (see Figure 27) to connect to the ELAN and TLAN.

Figure 27
VGMC Adaptor



NTDU15 Media Gateway Expansion chassis

Figure 28
Media Gateway Expansion

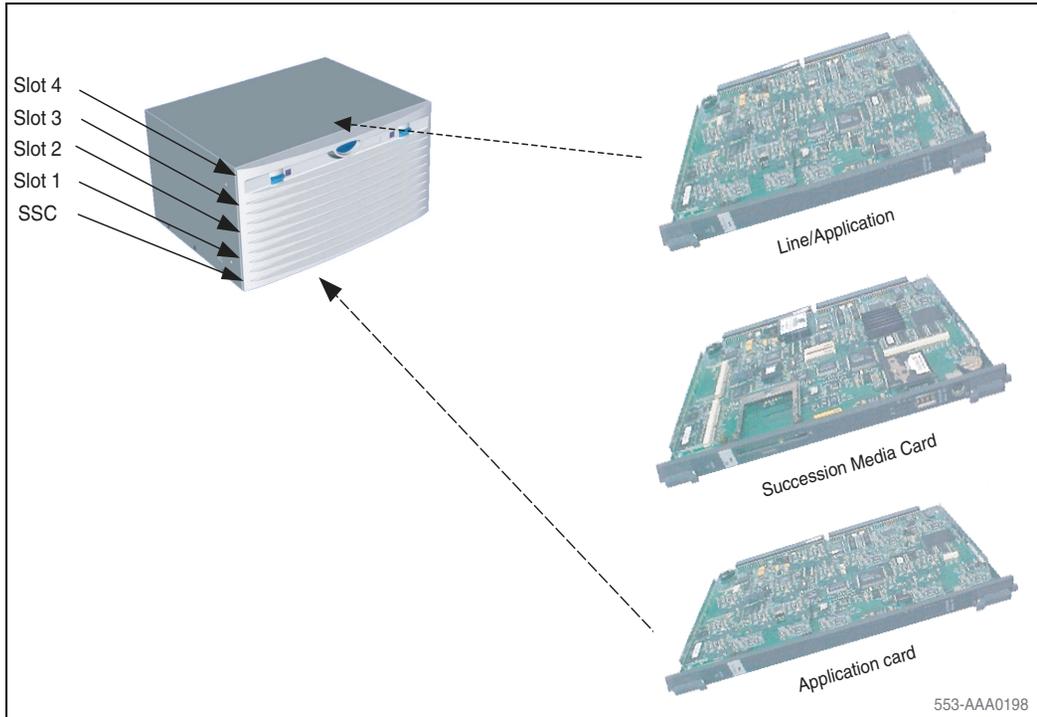


Figure 29
NTDU14 Media Gateway Expansion connectors

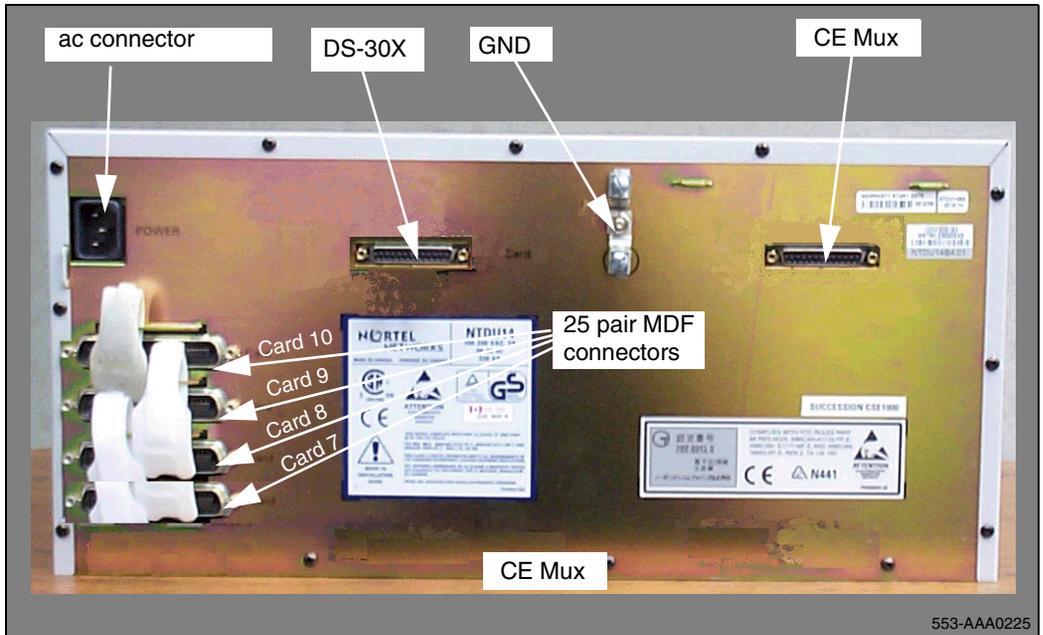


Figure 29 shows the following connectors:

- The ac power cord connector that provides an ac connection to the Media Gateway Expansion.
- GND for ground cable termination.
- DS-30X used to interconnect the Media Gateway and Media Gateway Expansion Peripheral Equipment (card) bus.
- CE Mux used to interconnect the Media Gateway and Media Gateway Expansion Peripheral Equipment (card) bus.
- 25-pair connectors used to extend PE card data to the MDF.

Media Gateway/Expansion card slot assignment

The Media Gateway and Media Gateway Expansion contain physical card slots numbered 1 to 10. When configuring the Succession 1000 system, you must transpose the physical card slot numbers to “logical” card slot numbers. For example, to configure a card physically located in slot 2 of the first Media Gateway, use logical slot 12. To configure a card physically located in slot 2 of the second Media Gateway, use logical slot 22. See Table 2 on [page 82](#).

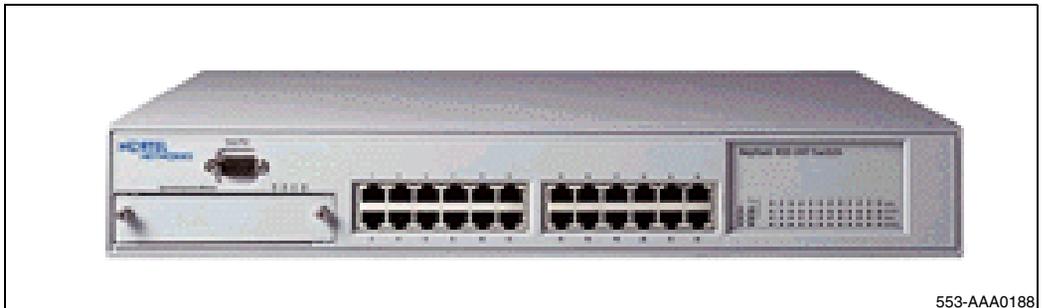
Table 2
Media Gateway and Media Gateway Expansion slot assignments

		Media Gateway/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
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	*	
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. Note: The bottom card slot in the Media Gateway chassis is reserved for the SSC card.									

Ethernet switch (customer-supplied)

The customer-supplied Layer 2 Ethernet switch transmits data packets to interconnected Ethernet-attached devices. The switch directs the data only to the targeted device, rather than to all attached devices. See Figure 30 on [page 83](#). See *Data Networking for Voice over IP* (553-3001-160) for details.

Figure 30
BayStack 450



Power over LAN (optional)

An optional Power over LAN unit adds power and data communication over standard Category 5 LAN drops for powering Internet Telephones. The Power over LAN unit eliminates the need to connect each telephone to an ac power outlet. This saves in desktop wiring and enables the use of a centralized Uninterruptible Power Supply (UPS) for power backups. Using an Power over LAN unit eliminates the need to use a separate power transformer for each Internet Telephone.

The Power over LAN unit shown in Figure 31 on [page 84](#) has 24 powered ports, and one panel can power 24 Internet Telephones. Each port has a data input jack and a data plus power output jack for a total of 48 jacks. All jacks are RJ-45 type. There is a fault lamp for each port.

Note: Some Internet Telephones require a power splitter, while other Internet Telephones have a power splitter built-in.

The Power over LAN unit is patched in between an Ethernet switch and the individual data drops. See Figure 32 on page 85. The Ethernet switch is patched to the data input jack and the data plus power jack is connected to the data drop. You would normally rack mount a Power over LAN unit in the wiring closet close to the switch but you can stack the Power over LAN unit.

Figure 31
Power over LAN unit

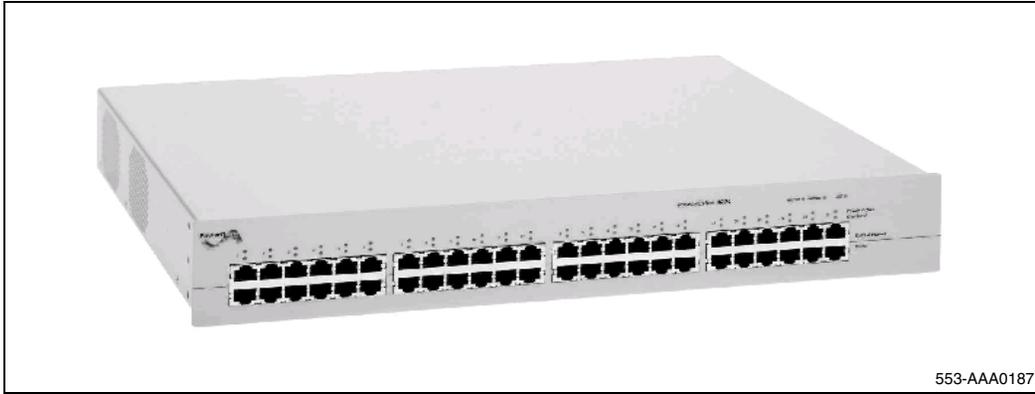
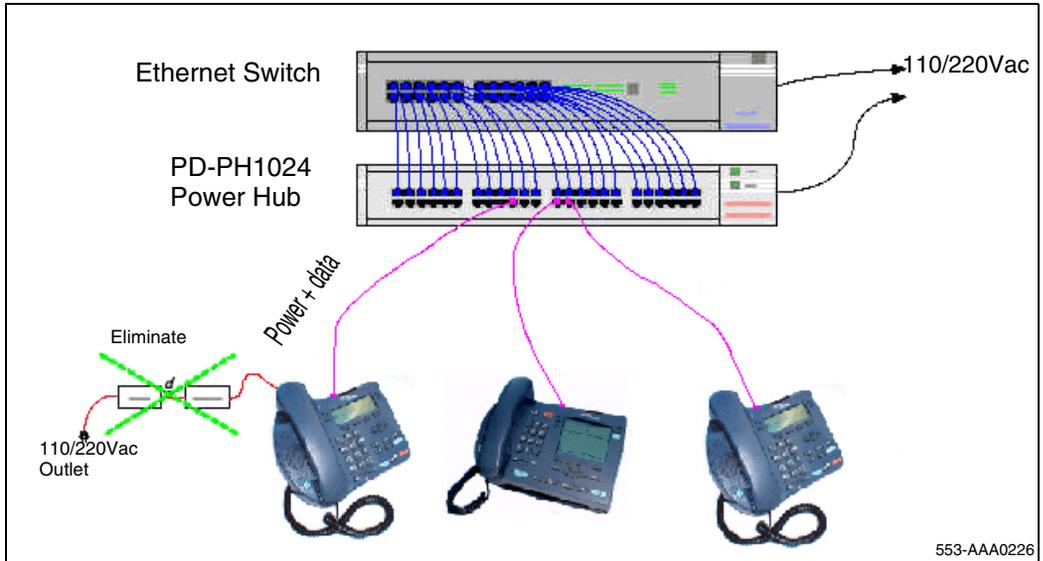


Figure 32
Power panel patch

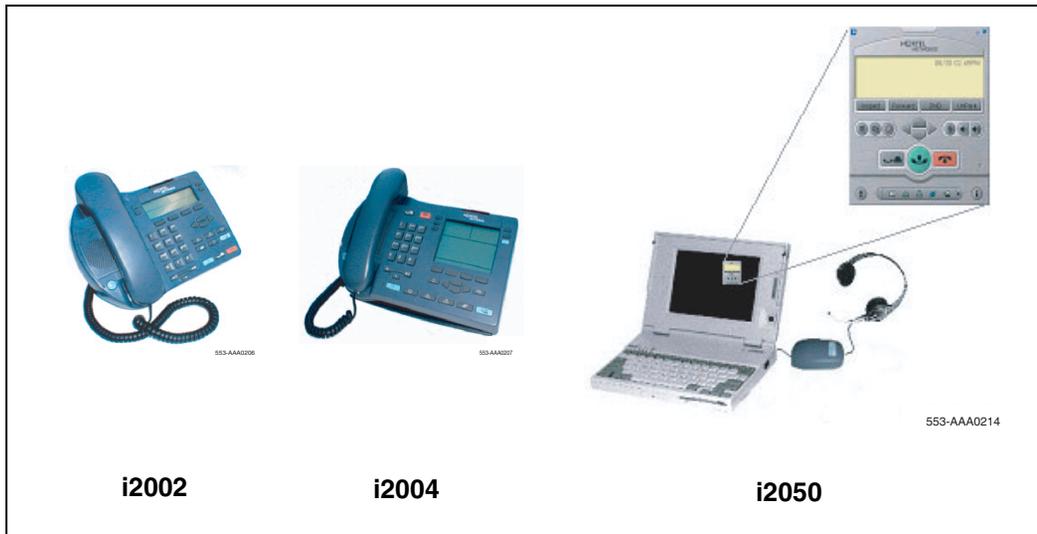


Internet Telephones

The Succession 1000 system can support the following Internet Telephones:

- i2002
- i2004
- i2050 SoftPhone

Figure 33
Internet telephones



i2004 Internet Telephones accessories

The i2004 Internet Telephones use the following accessories as shown in Figure 34 on [page 87](#):

- Internet Telephone Switch Module, provides a 3-port data switch to share a single LAN drop (only required for earlier versions of the i2004 Internet Telephone)
- Power Splitter, used with power over the LAN

Note: The i2002 Internet Telephone does not require these accessories.

Figure 34
i2004 Internet Telephone accessories

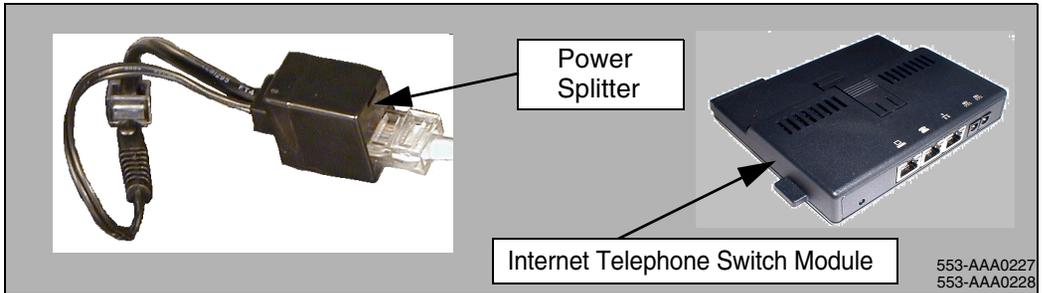


Figure 35
Internet Telephone Switch Module



Internet Telephone configuration

You can configure the Internet Telephones through a Dynamic Host Configuration Protocol (DHCP) server. See *IP Line: Description, Installation, and Operation* (553-3001-365) for details.

NTVQ01 Succession Media Card

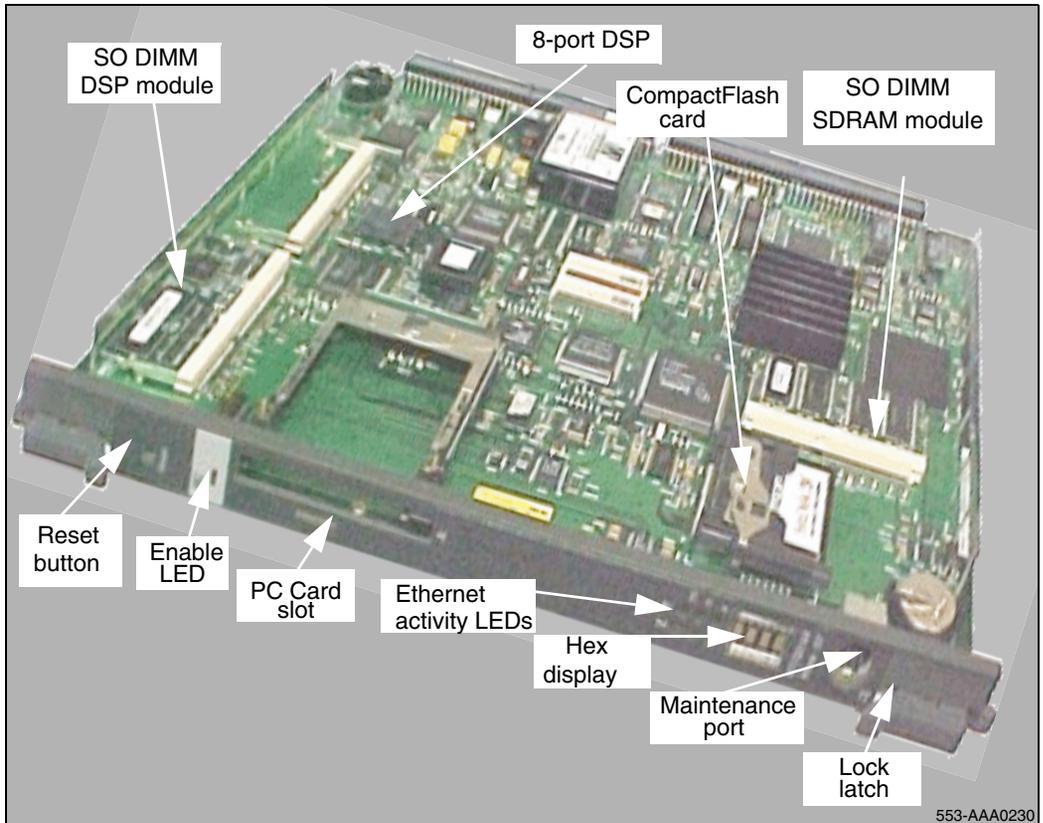
The NTVQ01 Succession Media Card has both faceplate features, and backplane interfaces which are used to connect to an external LAN. This section provides information on the faceplate connectors and indicators.

Note: The Media Card can run various applications. For example, a Media Card with the IP Line 3.0 application is referred to as a Voice Gateway Media Card (VGMC). The Media Card with the MIRAN III application is referred to as a MIRAN III Media Card.

Faceplate connectors and indicators

Figure 36 on [page 89](#) shows the NTVQ01 Media Card's faceplate connectors and indicators.

Figure 36
Succession Media Card



553-AAA0230

Reset button

Use the reset switch on the faceplate to manually reset the Media Card.

Status LED

The Media Card faceplate red LED indicates the following:

- the enabled/disabled status of the card
- the self-testing result during power up or card insertion into an operational system

PC Card slot

The PC Card slot accepts standard PC Card flash cards, including ATA Flash cards (3 Mbit/s to 170 Mbit/s).

Nortel Networks supplies PC Card adaptors you can use in the CompactFlash slot.

This slot is used (depending on the card's application) for Media Card software upgrades, backing up announcements, and additional storage.

Ethernet activity LEDs

The Media Card faceplate contains Ethernet activity LEDs for each network.

Maintenance hex display

A four-digit LCD-based hexadecimal display provides the following information:

- **T:xx** messages indicating the status of the internal self-test during boot-up
- **xxxx** indicating that an application is running on the card

RS-232 Asynchronous Maintenance Port

An 8-pin mini-DIN socket on the Media Card faceplate provides access to the RS-232 port. The maintenance faceplate port can provide access to the Media Card for OA&M purposes. This port is also available via a female DB9 connector on the 50-pin I/O Adaptor. You should use the DB9 connector to make a permanent terminal connection.

8-port DSP

The 8-port Digital Signal Processor (DSP) is a non-replaceable component providing eight channels: ports 0 to 7.

SO DIMM DSP

The Small Outline Dual Inline Memory Module Digital Signal Processor is an expansion component that provides twenty four ports: ports 8 to 31.

SO DIMM SDRAM

The Small Outline (SO) Dual Inline Memory Module (DIMM) Synchronous Dynamic Random Access Memory (SDRAM) provides memory for the IP Line 3.0 application and other applications.

CompactFlash card

The CompactFlash card contains the IP Line 3.0 application.

IP Line 3.0 application

When the IP Line 3.0 application runs on a Media Card, that card is a Voice Gateway Media Card (VGMC). The IP Line 3.0 application has the following components:

- Terminal Proxy Server (TPS)
- Voice Gateway

Terminal Proxy Server

Internet Telephones can register to the TPS on this card if there is no Signaling Server in the IP Telephony node. If there is a Signaling Server, the Internet telephones register to the Signaling Server TPS.

The TPS acts as an Internet Telephone line card.

Voice Gateway Media Card

The Voice Gateway Media Cards (VGMCs) are used any time an IP and TDM device are connected together. A VGMC provides Digital Signal Processor (DSP) ports for accessing TDM resources. The physical TNs are the gateway channels (DSP ports).

Voice Gateway Media Card Hex display

Lxxx indicates the VGMC is running active leader tasks, where xxx = number of Internet Telephones registered on the card.

Fxxx indicates the VGMC has detected the active leader, and is running Follower tasks, where xxx = number of Internet Telephones registered on the card.

Mxxx indicates that the VGMC has become the node master when the previous leader became unavailable.

See *IP Line: Description, Installation, and Operation* (553-3001-365) for additional VGMC Hex display codes.

Software architecture

The following topics describe the software required for the Succession 1000 system.

Call Server and Media Gateway software

The Call Server and Media Gateway software is stored on the SSC Software Daughterboards.

Either a pre-programmed SSC Software Daughterboard can deliver the software or a software delivery card (PC card) can install software on the SSC Software Daughterboard. A software delivery card can also upgrade the software. The software includes the boot-ROM and IP Daughterboard firmware.

Signaling software

The Signaling Server software is stored on the Signaling Server hard disk. The software is delivered on a CD-ROM. You can also upgrade the software with a CD-ROM.

Voice Gateway Media Card loadware

The Voice Gateway Media Card IP Line 3.0 loadware is stored on the CompactFlash card. The IP Line 3.0 loadware comes pre-installed on the CompactFlash card. You can upgrade the IP Line 3.0 loadware either with Element Management or with a PC card inserted into the faceplate.

The IP Line 3.0 loadware is delivered on the Signaling Server CD-ROM. The VGMC loadware files are downloaded using Element Management, and the files are retrieved from the Signaling Server. (The loadware files are copied from the CD-ROM to the Signaling Server hard disk during the Signaling Server installer upgrade.)

i2002 and i2004 Internet Telephone firmware

The Internet Telephone firmware is stored in the Internet Telephone flash memory. The firmware comes pre-installed in the flash memory. You can deliver the firmware from the Signaling Server, if the firmware is required when the Internet Telephone registers. (The firmware files are copied from the CD-ROM to the Signaling Server hard disk during the Signaling Server installer upgrade.)

i2050 Internet Telephone application

The i2050 Internet Telephone is a Windows application.

Software delivery

Software is delivered to the system components using the following format:

- Download the software from the Nortel Networks Software Download web site to the management workstation.
- Upload the software from the management workstation to the Signaling Server using the “Centralized file upload” page in Element Manager.
- Distribute the software to the IP telephony components (such as the Signaling Server, Voice Gateway Media Cards, and Internet Telephones) using Element Manager.

Centralized Automatic Software Upgrade

Centralized Automatic Software Upgrade enables loading a new version of software automatically to the Media Gateways after the Call Server software is upgraded. To reduce service impact, a sequential upgrade mode upgrades only one Media Gateway at a time.

Centralized upgrade

Centralized upgrade enables the IP telephony components (VGMCs) to be upgraded from the Element Manager interface. The Element Manager host, the Signaling Server, is a file server for software upgrade files for the VGMCs. Internet Telephones also automatically upgrade from a centralized file location (their TPS).

Centralized patching

Centralized patching enables IP telephony components, the Call Server, and Media Gateways, to be patched from a central location through Element Manager.

File uploading

File uploading enables software upgrade files and patches to be uploaded to the Element Manager host, the Signaling Server, for centralized upgrading or centralized patching. The file is uploaded from the management PC (web browser) to the Element Manager host (web server).

Patching implementation

The Call Server, Media Gateways, Signaling Server, and the VGMCs are patchable. Installing a patch enables a fix to be delivered, without requiring a new version of software.

System management

System Management follows the Faults Configuration Accounting Performance Security (FCAPS) methodology. For details on System Management see *System Management* (553-3001-300).

Element Manager

Element Manager provides the ability to configure and maintain certain aspects of the Succession 1000 system through a web interface. Single web pages provide access to information traditionally spread into multiple overlays.

Element Manager provides the tools required to configure and maintain the following components of the Succession 1000 system:

- Succession 1000 Call Server
- Succession 1000 Media and Branch Office Gateways
- Signaling Server
- IP Line 3.0 / Voice Gateway

Call Server management

The system management platform for Succession 1000 requires that the Call Server be configured using a web-based user interface.

You can configure and manage the Call Server with Element Management for following system information:

- Configuration Record
- Customer Data Block
- Route Data Blocks
- Trunks
- ESN Data Blocks

Note: Element Manager does not provide a tool for managing telephones. For information on configuring and managing the telephones in a Succession 1000 system, see the Station Administration chapter in *Optivity Telephony Manager: System Administration* (553-3001-330).

Media Gateway management

The Media Gateway is configured to operate as a standalone PBX using trunk and analog line interfaces.

You can configure and manage the Media Gateway with Element Manager for following system information:

- Configuration record
- Customer data block
- Route data blocks
- Trunks
- ESN data block

To view a complete list of the Branch Office Gateway parameters that can be configured and managed using Element Manager, see *Branch Office (553-3001-214)*.

Signaling Server management

Installation of the Signaling Server is performed using a CD-ROM and a PC or terminal connected to a serial port. After installation of the software and configuration of basic information about the Signaling Server, such as IP information, you can configure the Signaling Server components using the web-based interface. The web server is installed on each Signaling Server in a Succession 1000 system. All HTML Web pages and data files required for web-based Element Manager of the Call Server, Signaling Server, and Voice Gateway Media Cards are installed on the Signaling Server as part of the standard installation on upgrade process.

Voice Gateway Media Card management

Element Manager provides the tools to configure and maintain the following components of the Voice Gateway Media Card:

- IP addresses
- SNMP related data
- DSP parameters

You can view configuration details and management parameters on the Voice Gateway Media Card using Element Manager. See *IP Line: Description, Installation, and Operation* (553-3001-365).

The following maintenance activities are supported using Element Manager for the Voice Gateway Media Card:

- enable/disable and reset the VGMC
- perform maintenance to the VGMC
- download new software for upgrades
- gather Operational Measurement (OM) reports

Note: Configuration data for the Voice Gateway Media Card is stored on the Call Server, where it is backed up and restored along with the Call Server configuration data.

Call Server configuration

You can configure the Call Server with either Command Line Interface (CLI) or Element Manager for the following overlays/loads:

- LD 02 – Traffic
- LD 14 – Trunk Data Block
- LD 15 – Customer Data Block
- LD 16 – Route Data Block
- LD 17 – Configuration Record 1
- LD 20 – LD 22 - Print Reports

- LD 32 – Network and Peripheral Equipment Diagnostic
- LD 36 – Trunk diagnostic
- LD 43 – Equipment datadump
- LD 49 – New Flexible Code Restriction and Incoming Digit Conversion
- LD 60 – Digital Trunk Interface and Primary Rate Interface
- LD 73 – Digital Trunk Interface
- LD 86, LD 87, LD 90 – Electronic Switched Network
- LD 96 – D-Channel Diagnostic

The following components are used for Web-based administration of the overlays described in the preceding section:

- interface to communicate with the Web Server
- database transaction server to process XML files
- database interface

LD 21 is modified to provide easier navigation to the physical elements and software configuration components.

Online Help supports configuration for the Call Server, Media and Branch Office Gateways, Signaling Server and Voice Gateway Media Cards. Configuration and maintenance error messages are displayed.

Reliability strategies

Contents

This section contains information on the following topics:

Overview	100
Component redundancy	100
Call Server redundancy	101
Alternate Call Server	101
Signaling Server redundancy	101
H.323 Gatekeeper redundancy	102
Campus distributed Media Gateway in Survival Mode	104

Overview

Communications reliability is critical to the operation of any business. A number of capabilities are available in Succession 1000 to ensure that telephony is available when one or more of the following situations occur:

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

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

- Call Server with automatic database distribution
- Signaling Server software, including H.323 Gateway and Internet Telephone software
- H.323 Gatekeeper
- H.323 Gateway interface to Gatekeeper
- Campus-distributed Media Gateway in Survival Mode

Component redundancy

You can provision a Succession 1000 to provide the following component redundancy:

- If the Call Server is unavailable, you can configure Media Gateways as survivable independent systems (automatic database synchronization).
Note: If the Media Gateway is not configured as survivable, then it is out of service until the Call Server is in service again.
- If the Call Server is unavailable, the IP telephony node can re-register with one Media Gateway specified as an Alternate Call Server.
- If the master (Leader) Signaling Server is unavailable, a Follower Signaling Server becomes the IP Telephony node master.
- Signaling Servers share set registrations (load-sharing).

- Virtual trunk redundancy. If a virtual trunk is unavailable, the call-processing software selects an alternate route.
- If the Primary Gatekeeper is unavailable, you can configure an Alternate Gatekeeper (automatic database synchronization).
- If the Primary and Alternate Gatekeeper are unavailable, you can configure a Failsafe Gatekeeper (at each Signaling Server).

Call Server redundancy

All Media Gateways are equipped with a full set of call-processing software components. The Media Gateways maintain a configuration database that is periodically synchronized with the Call Server. During normal operation, the processor in the Media Gateway handles low-level control of the interface cards installed in the Media Gateway slots. The processor communicates with the Call Server for feature operation. If the Media Gateway processor loses communication with the Call Server (due to failure of the Call Server, or IP network components such as cabling and L2 switch) the Media Gateways assume Call Server responsibilities for all Media Gateway hardware and accessible Internet Telephones.

Alternate Call Server

You can configure a Media Gateway SSC as a survivable Alternate Call Server. An IP Telephony node can register with an Alternate Call Server, if the primary Call Server is unavailable. For more information, see *IP Line: Description, Installation, and Operation* (553-3001-365) and *Succession 1000 System: Planning and Engineering* (553-3031-120).

Signaling Server redundancy

Redundancy of the Signaling Server is provided on a load-sharing basis for Internet Telephone Proxy Server (TPS) and as an alternate route for the H.323 Media Gateway software.

In the unlikely event of the Leader Signaling Server failure, a Follower Signaling Server becomes the new IP Telephony node master. If there are no Follower Signaling Servers, the Internet Telephones can register with the TPS on the Voice Media Gateway Card (VGMC). For further information on Leader Signaling Server failure, refer to *IP Peer Networking* (553-3001-213).

H.323 Gatekeeper redundancy

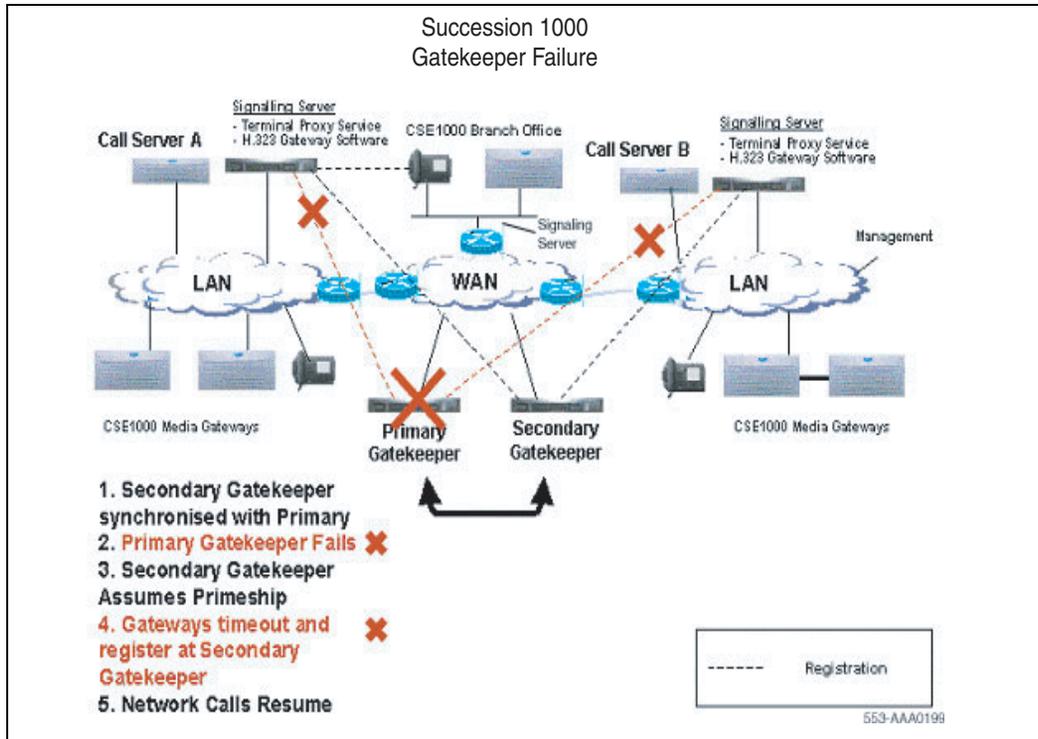
The Succession 1000 networks are equipped with at least one H.323 Gatekeeper to provide management of the network numbering plan for private and public numbers. You can install an optional redundant Gatekeeper in the network. This Alternate Gatekeeper automatically synchronizes its database with the Primary Gatekeeper periodically.

The H.323 Signaling Gateway (virtual trunk) software of each Succession 1000 runs on the Signaling Server. The software is aware of both a Primary and (optional) Alternate Gatekeeper. If this virtual trunk software loses communication with its Primary Gatekeeper, it automatically registers with the Alternate Gatekeeper and continues to operate.

A “Keep-Alive” timer ensures that if a Gatekeeper stops responding for a specified amount of time, the virtual trunk software registers at the Alternate Gatekeeper to resume operation.

Figure 37 on [page 103](#) illustrates the handling of the H.323 Signaling Gateway (virtual trunk) interface and Alternate Gatekeeper in the event of a Primary Gatekeeper failure.

Figure 37
Gatekeeper failure – Alternate (secondary) Gatekeeper



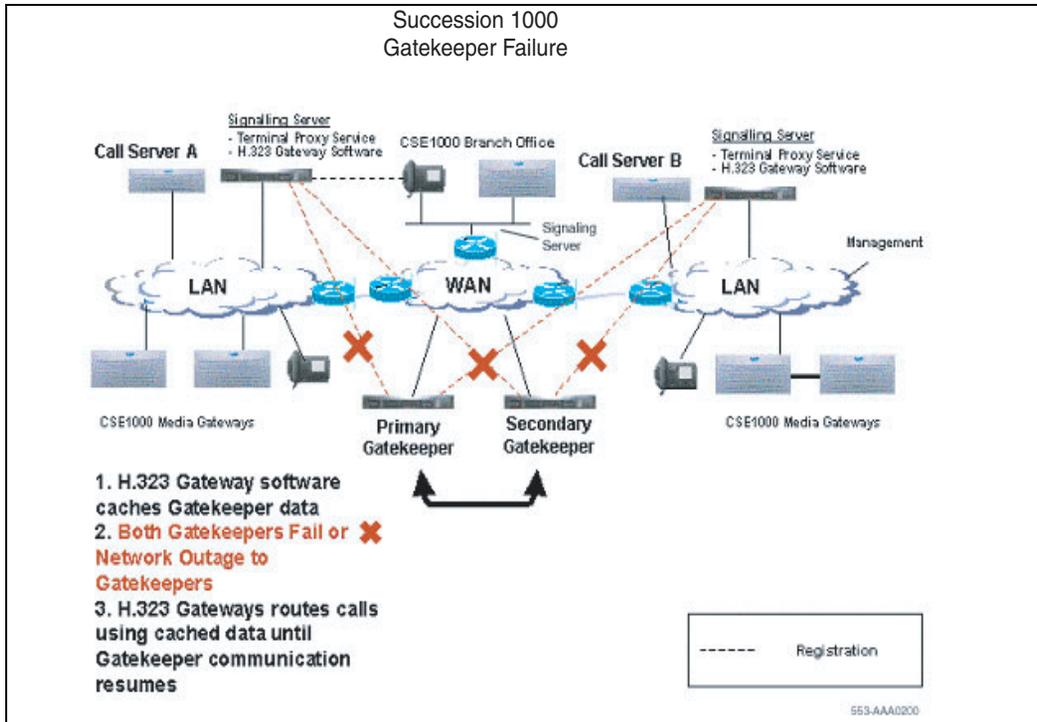
Failsafe Gatekeeper

In addition to Gatekeeper redundancy, you can configure the H.323 Signaling Gateway (virtual trunk software) to act as a Failsafe Gatekeeper. As a Failsafe Gatekeeper, the H.323 Signaling Gateway interfaces can withstand loss of communication to both Gatekeepers. The H.323 Signaling Gateway reverts to a locally-cached copy of the Signaling Gateway addressing information.

Calls are routed using the cached data. Since this cache is static (not updated) until one of the Gatekeepers becomes accessible, it is intended to be used only for a brief network outage.

The Failsafe Gatekeeper is illustrated in Figure 38 on [page 104](#).

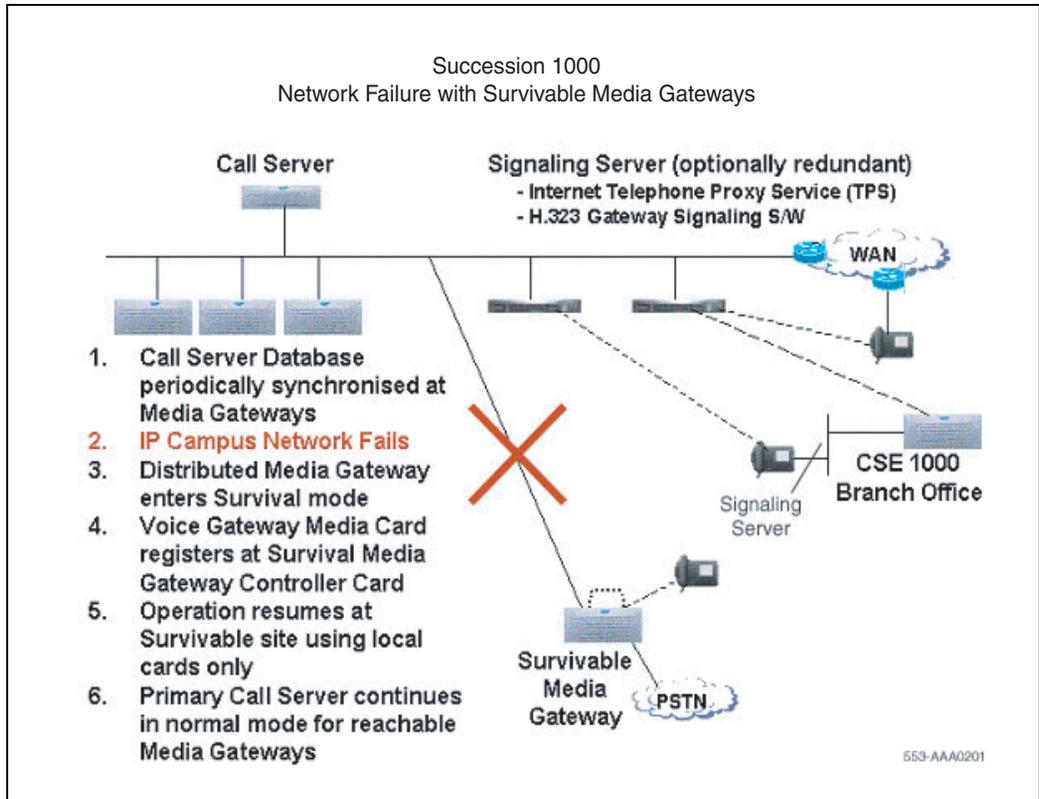
Figure 38
Gatekeeper failure – Failsafe Gatekeeper



Campus distributed Media Gateway in Survival Mode

You can configure Succession 1000 Media Gateways as survivable when distributed throughout a campus environment. This provide basic telephony services in the event of a network outage. Figure 39 on [page 105](#) illustrates the handling of such an outage.

Figure 39
Network failure



See *Succession 1000 System: Planning and Engineering* (553-3031-120) for details on Alternate Call Server and survivability.

Succession 1000

Succession 1000 System

Overview

Copyright © 2003 Nortel Networks

All Rights Reserved

Information is subject to change without notice. Nortel Networks reserves the right to make changes in design or components as progress in engineering and manufacturing may warrant. This equipment has been tested and found to comply with the limits for a Class A digital device pursuant to Part 15 of the FCC rules, and the radio interference regulations of Industry Canada. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy, and if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference in which case the user will be required to correct the interference at their own expense.

SL-1, Meridian 1, and Succession are trademarks of Nortel Networks.

Publication number: 553-3031-010

Document release: Standard 1.00

Date: October 2003

Produced in Canada

NORTEL
NETWORKS™