System Redundancy Fundamentals
Avaya Communication Server 1000

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Chapter 1: New in this release

The following sections detail what's new in *Avaya System Redundancy Fundamentals, NN43001-507* for Avaya Communication Server 1000:

- Features on page 11
- Other changes on page 11

Features

There are no updates to the feature descriptions in this document.

Other changes

Revision history

April 2013	Standard 05.09. This document is up-issued to include USB as a Backup Rule type.
February 2013	Standard 05.08. This document is up-issued to include updates to the Survivable SIP Media Gateway chapter.
June 2012	Standard 05.07. This document is up-issued to include updates to the sections for configuring a Survivable Media Gateway and configuring a Backup Rule.
May 2012	Standard 05.06. This document is up-issued to include updates to the Database replication section.
August 2011	Standard 05.05. This document is up-issued to support Communication Server 1000 Release 7.5.
April 2011	Standard 05.04. This document is up-issued to support Communication Server 1000 Release 7.5.
March 2011	Standard 05.03. This document is up-issued to include updates to the section for adding a Secondary Call Server.

November 2010	Standard 05.01 and 05.02. This document is up-issued for Avaya Communication Server 1000 Release 7.5.
June 2010	Standard 04.02. This document is up-issued to include information on Survivable SIP Media Gateway.
May 2010	Standard 04.01. This document is up-issued for Avaya Communication Server 1000 Release 7.0.
June 2009	Standard 03.02 and 03.03. This document is up-issued for Communication Server 1000 Release 6.0.
May 2009	Standard 03.01. This document is up-issued for Communication Server 1000 Release 6.0.
July 2008	Standard 02.04. This document is up-issued to include changes to technical content.
April 2008	Standard 02.03. This document is up-issued to include changes to technical content.
February 2008	Standard 02.02. This document is up-issued to include changes to technical content.
December 2007	Standard 02.01. This document is up-issued for Communication Server Release 5.5.
June 2007	Standard 01.02. This document is up-issued to include changes to the procedure for database restoration.
May 2007	Standard 01.01. This document This document is renumbered and updated for Communication Server 1000 Release 5.0. This document contains information previously contained in the following legacy document, now retired: CS 1000: System Redundancy (553-3001-307).
July 2006	Standard 4.00. This document is up-issued to include recommendation that GRSEC package (405) be equipped during the software installation process.
April 2006	Standard 3.00. This document is up-issued to include recommendation about enabling a switch of the CPP core processors during Daily Maintenance Routines (DROL).
August 2005	Standard 2.00. This document is issued to support Communication Server 1000 Release 4.5. This version contains numerous changes to campus redundancy.
September 2004	Standard 1.00. This document is issued to support Communication Server 1000 Release 4.0. Element Manager screens, Automatic NUID creation, and NRS routing are incorporated.

Note on legacy products and releases

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

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Applicable systems

This document applies to the following systems:

- Avaya Communication Server 1000M Single Group (Avaya CS 1000M SG)
- Avaya Communication Server 1000M Multi Group (Avaya CS 1000M MG)
- Avaya Communication Server 1000E (Avaya CS 1000E)

Conventions

Terminology

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

- Avaya Communication Server 1000E (Avaya CS 1000M) CP PIV, CP PM
- Avaya Communication Server 1000M (Avaya CS 1000E)
- Meridian 1

System Manager and Session Manager

In this document, the following terms apply:

- On systems where System Manager is available, the term UCM in the documentation refers to UCM in System Manager. On systems where System Manager is not available, the term UCM in the documentation remains unchanged.
- On systems where System Manager 6.2 is available, the term Subscriber Manager in the documentation refers to User Profile Management in System Manager. On systems where System Manager 6.1 is available, the term Subscriber Manager refers to

Subscriber Manager in System Manager. On systems where System Manager is not available, the term Subscriber Manager in the documentation remains unchanged.

- On systems where Session Manager is available, the term NRS in the documentation refers to Session Manager; on systems where Session Manager is not available, the term NRS in the documentation remains unchanged.

Related information

This section lists information sources that relate to this document.

Documents

The following documents are referenced in this document:

- *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*
- *Avaya Dialing Plans Reference, NN43001-283*
- *Avaya Network Routing Service Fundamentals, NN43001-130*
- *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*
- *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*
- *Avaya Branch Office Installation and Commissioning, NN43001-314*
- *Avaya Features and Services Fundamentals, NN43001-106*
- *Avaya Software Input/Output Administration, NN43001-611*
- *Avaya Software Input/Output Reference Maintenance, NN43001-711*
- *Avaya Communication Server 1000E Planning and Engineering, NN43041-220*
- *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*
- *Avaya Security Management Fundamentals, NN43001-604*
- *Avaya Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*
- *Avaya SIP Line Fundamentals, NN43001-508*

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- Getting product training on page 15
- Getting help from a distributor or reseller on page 15
- Getting technical support from the Avaya Web site on page 16

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Chapter 3: Overview

Contents

This section contains information about the following topics:

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Geographic Redundancy

Avaya Communication Server 1000E (Avaya CS 1000E) (CP PIV, CP PM) and Avaya Communication Server 1000M (Avaya CS 1000M) (CP PIV) systems provide core redundancy using dual processors to ensure a system remains operational following a local component failure.

Geographic Redundancy further increases the reliability of Avaya CS 1000E and Avaya CS 1000M systems by providing a remote system to serve as a backup for a local system. Depending on the configuration, the remote backup system ensures continued service for resources in the event of a catastrophic failure, such as damage to equipment caused by floods or fire.

Geographic Redundancy provides several flexible configurations to achieve the required reliability and careful planning will determine the correct solution for each installation.

Geographic Redundancy is used to describe configurations that include combinations of the following sub-components.

Multiple redundant Call Servers

Depending on the geographic organization and desired level of redundancy, a system can be configured to include 1 or up to 50 Secondary redundant Call Servers. Secondary Call Servers can be configured at each individual location or grouped into regional redundancy clusters. They can also include redundant call processor pairs.

Database replication from the Primary to Secondary Call Servers

The Geographic Redundancy model is based on a central Primary Call Server with a single customer database. The Primary site database is replicated to all Secondary redundant Call Servers. All Secondary Call Servers have a complete customer database image and can take over support for all or a subset of the Primary system IP Phones and Survivable Media Gateways.

For more information about database replication, see Database replication on page 31.

IP Phone redirection

During normal operation, Geographic Redundancy redirects IP Phones to the Primary site while allowing the Secondary sites to assume and maintain support for the IP Phones if the Primary site is not reachable. The NRS guides the Secondary site in the redirection of IP Phones to the Primary site.

For more information about IP Phone redirection, see Geographic Redundancy IP Phone redirection on page 93.

Survivable Media Gateway

Survivable Media Gateway is a component of Geographic Redundancy allowing a Media Gateway to register to up to 2 Alternate Call Servers (AC1 and AC2) if the Primary Call Server is not reachable. The Alternate Call Servers are directly configured for each Media Gateway. Survivable Media Gateways in a system can have different Secondary Call Servers depending on geographic requirements.

For more information about Survivable Media Gateway, see Geographic Redundancy Survivable Media Gateway configuration on page 23.

Note:

Avaya no longer recommends Geographic Redundancy 1+1 configurations. 1+1 configurations are still supported in Release 6.0; however, no new packages are available. For information about 1+1 configurations, refer to previous issues of this document.

Survivable SIP Media Gateway

Avaya Communication Server 1000E Survivable SIP Media Gateway (Survivable SIP MG) is a deployment option that enhances Geographic Redundancy (GR) by supporting up to 511 Survivable SIP Media Gateways.

A Survivable SIP Media Gateway is similar to the traditional Survivable Media Gateway (SMG) for which the Survivable Call Server receives the database from the Primary Call Server and all IP Phones are redirected to the Primary Call Server during normal operations. However, unlike the SMG, an additional blade called a SIP Media Gateway Controller is used. The Media Gateway Card is always registered to the SIP Media Gateway Controller in normal and survival modes. All TDM devices, such as telephones and trunks, are configured on the SIP Media Gateway Controller.

The Survivable SIP Media Gateway consists of two components:

- SIP Media Gateway
- Survivable Call Server

SIP Media Gateway

For the SIP Media Gateway the choice of hardware is dictated by the number of users at the location which the Survival SIP Media Gateway serves. For a typical location, the CP DC card is recommended as the Survivable Server platform and the CP MG card is recommended for the SIP Media Gateway. The CP DC and the CP MG typically run the co-resident Call Server and Signaling Server applications.

Survivable Call Server

The Survivable Call Server is a Common Processor running the Co-resident Call Server and Signaling Server software with a replicated copy of the Primary Call Server database. The Survivable Call Server functions as the GR Secondary Call Server during network outages.

In a Survivable SIP MG configuration, the Media Gateways do not register to Alternate Call Servers (AC1 and AC2) but rather to their local SIP Media Gateway, which is distinct from the Secondary Call Server. The Secondary Call Server is a Survivable Call Server that provides redundancy if there is a loss of connectivity to the Primary Call Server (PCS).

A SIP Media Gateway configured as survivable enables the delivery of a fully featured network based survivable service for IP clients. The survivability of the IP clients is automatically configured from the PCS and any Moves, Adds and Changes (MACs) are automatically downloaded to the Survivable Call Server, also known as the Secondary Call Server, thereby ensuring continuous database alignment.

The SIP Media Gateway is a local component with its own database and is not involved in the Geographic Redundancy database replication operation.

Note:

In a Survivable SIP MG configuration, you configure the database for the Survival Call Server on the PCS and then replicate it to the SCS. The database for the SIP Media Gateway is configured independently and is not related to the database configured on the PCS. Therefore, you should back up the database on the SIP Media Gateway independently of the PCS and SCS.

Deployment options

Avaya CS 1000E hardware provides several Survivable SIP MG deployment options. The number of cards required for each option is shown in the following table.

Table 1: Survivable SIP MG deployment options and hardware requirements

Deployment	Required Hardware				
	MGC	CP MG	CP PM	CP DC	COTS2
Option 1	1		2		
Option 2	1		1	1	
Option 3	1		1		1
Option 4	1			2	
Option 5	1			1	1
Option 6	1				2
Option 7		1	1		
Option 8		1		1	
Option 9		1			1

For information about configuring Survivable SIP Media Gateway, see Survivable SIP Media Gateway on page 167

Replicated databases

System Redundancy is maintained through database replication. Redundant CS 1000 systems replicate databases as:

- Survivable Media Gateway: Survivable Blade has replicated database of the Primary Call Server (PCS).
- Survivable SIP Media Gateway: Soft switch has replicated database of PCS and SIP Media Gateway is always controlled by Branch Call Server.

Controlled Load-sharing configuration

The Controlled Load-sharing configuration provides additional system redundancy for IP Phones by allowing active systems in a network to provide redundancy for each other.

The Controlled Load-sharing configuration can be implemented using different system types. An Avaya CS 1000M system can back up an Avaya CS 1000E system, and an Avaya CS 1000E system can back up an Avaya CS 1000M system.

The Controlled Load-sharing configuration employs similar functionality to the Branch Office feature (for more information about Branch Office, see *Avaya Branch Office Installation and Commissioning, NN43001-314*). You must configure a Directory Number (DN) and Terminal Number (TN) for each IP Phone on the home system where the IP Phone ultimately registers. On the backup system, the IP Phone is also assigned a DN and TN. This ensures that the backup system can provide the necessary functionality if the home system fails.

On the backup system, each IP Phone has an assigned Network User ID (NUID) and Network Home TN (NHTN), similar to the Branch User ID (BUID) and Main Office TN (MOTN). The NHTN corresponds to a home system TN and the NUID corresponds to a dialable home system DN where the IP Phone ultimately registers. The backup system uses the NUID value to redirect IP Phones to the home system for registration.

Each IP Phone in the network is configured with its Primary connect server (S1) pointing to its backup system. The backup system uses the defined NUID values to redirect each IP Phone to its home system for registration and normal operation. If the home system fails, the IP Phones remain registered on their backup system and receive service as normal.

The Temporary IP User license limits the number of IP Phone TNs which have NUID configured. The total number of license limitations for IP User, Basic IP User, or ACD agent usage does not include these TNs, which makes this option more cost-effective than Basic IP User or IP User licenses.

For more information, see Geographic Redundancy Controlled Load-sharing configuration on page 115.

N+1 configuration

Avaya does not support N+1 configurations. Existing N+1 configurations automatically convert to Controlled Load Sharing with Temporary IP User license configurations upon upgrade to Release 6.0 or later.

For information about N+1 configuration, refer to previous issues of this document.

Campus Redundancy

Campus Redundancy increases the redundancy of a single Avaya CS 1000E system through the physical separation of the Avaya CS 1000E High Availability (HA) Core Call Servers and allows the remote standby Call Server to assume system control if the active Call Server fails.

Note:

Avaya CS 1000M systems do not support Campus Redundancy.

You can use the Campus Redundancy feature to separate the Avaya CS 1000E Call Servers in a campus environment and enable two Call Servers, one active and one inactive, to connect through an Ethernet network interface. Campus Redundancy operates using a number of Layer 2 switching products, including the BayStack 470.

To separate the redundant Call Servers, the ELAN and HSP subnets can be extended between the two processors with Ethernet switches using Layer 2 protocol.

If the two Call Servers are collocated, you can connect them using a standard CAT5e or CAT6 crossover cable, limited to 100 meters in length.

For more information about Campus Redundancy, see Campus Redundancy on page 133.

Chapter 4: Geographic Redundancy Survivable Media Gateway configuration

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System monitoring on page 91

System redundancy task flow

This section provides a high-level task flow for the configuration of system redundancy. The task flow indicates the recommended sequence of events to follow when configuring a system and provides the document number that contains the detailed procedures required for the task.

For more information, refer to the following documents, which are referenced in the task flow diagram:

- *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*
- *Avaya Network Routing Service Fundamentals, NN43001-130*
- *Avaya Communication Server 1000E Software Upgrades, NN43041-458*

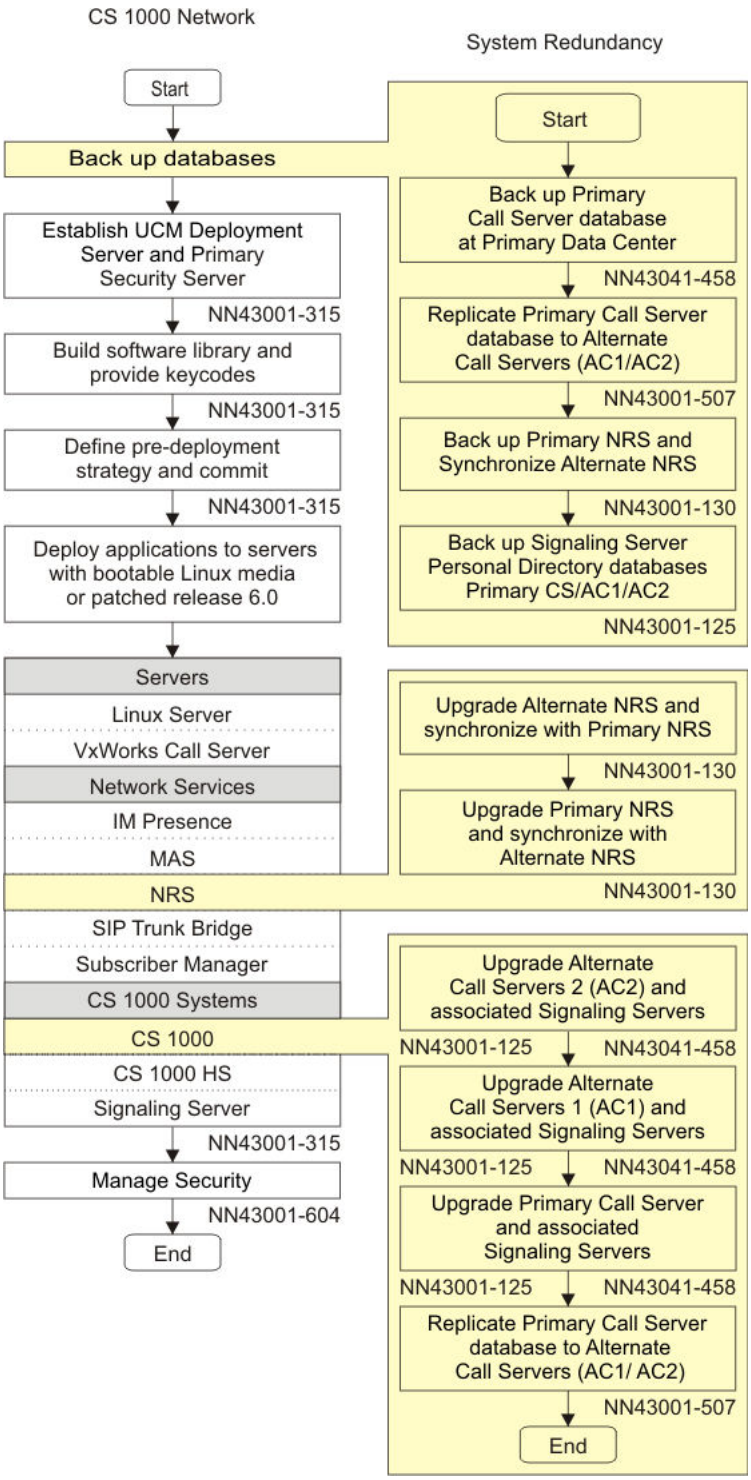

Figure 1: Task flow

Description

Survivable Media Gateway enhances the reliability of Avaya Communication Server 1000E (Avaya CS 1000E) systems by allowing the provisioning of up to 50 geographically remote Secondary Call Servers to a Primary Call Server. You can configure each Secondary Call Server as Alternate Call Server 1 or Alternate Call Server 2 for the devices assigned to it.

Survivable Media Gateway provides two levels of redundancy. If the Primary Call Server fails, local and remote resources register with the Secondary Call Server configured as Alternate Call Server 1. If the Primary and Secondary Call Servers both fail, or the WAN fails, local resources register with the Secondary Call Server that is installed at the local site and configured as Alternate Call Server 2.

In addition to the redundancy provided for IP resources, Survivable Media Gateway provides redundancy for TDM resources through triple IP registration. Media Gateways register to the Primary Call Server as first choice, Alternate Call Server 1 as second choice, and Alternate Call Server 2 as third choice. For more information about triple IP registration, see Triple IP registration on page 33.

In the Survivable SIP Media Gateway architecture the Media Gateway Controllers and all other media cards in the Media Gateway component register locally to their own call server.

The Primary Call Server administers the database, which is replicated to the Secondary Call Servers manually or by automatic scheduling. For more information about database replication, see Database replication on page 31.

You can implement the Survivable Media Gateway configuration using different system types. The Primary and Secondary Call Servers can be any combination of Avaya CS 1000E (CP PIV, CP PM) systems. Survivable Media Gateway can only be used with the Media Gateway Controller; it cannot be SSC based.

Figure 2: Survivable Media Gateway: Normal Configuration on page 27 shows a typical Survivable Media Gateway configuration.

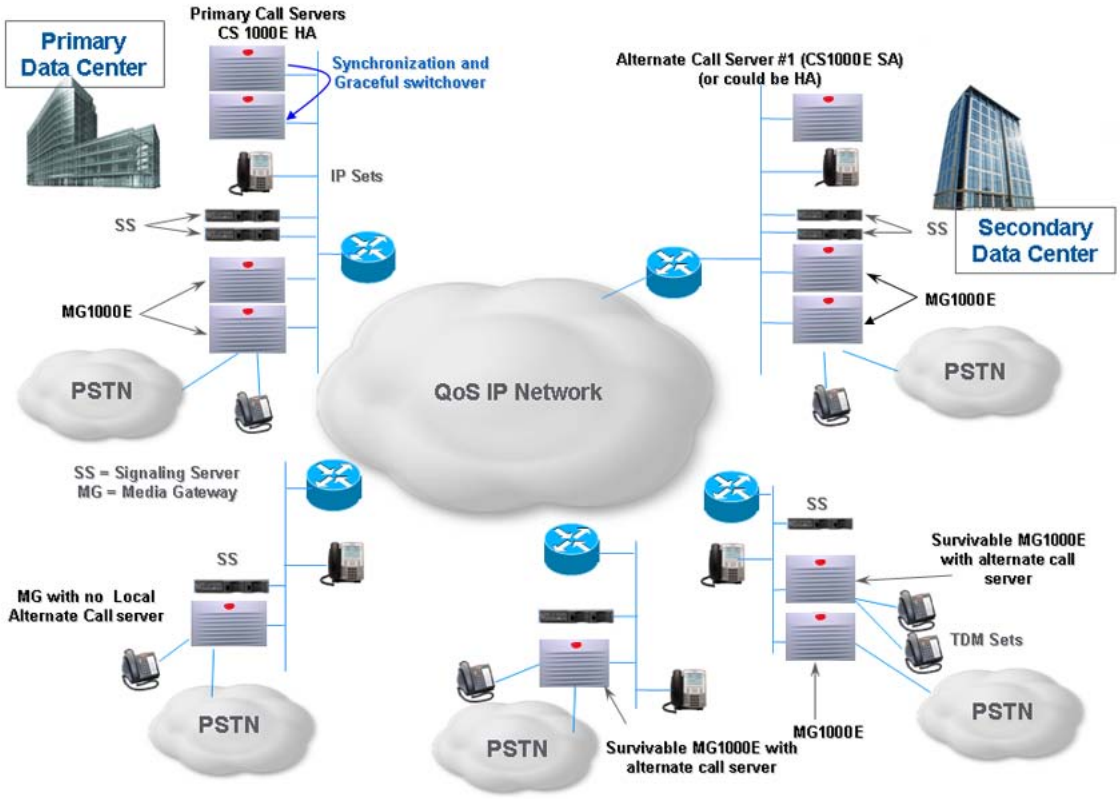

Figure 2: Survivable Media Gateway: Normal Configuration

Figure 3: Survivable Media Gateway: Database Replication on page 28 shows how the database replication process works during normal operations.

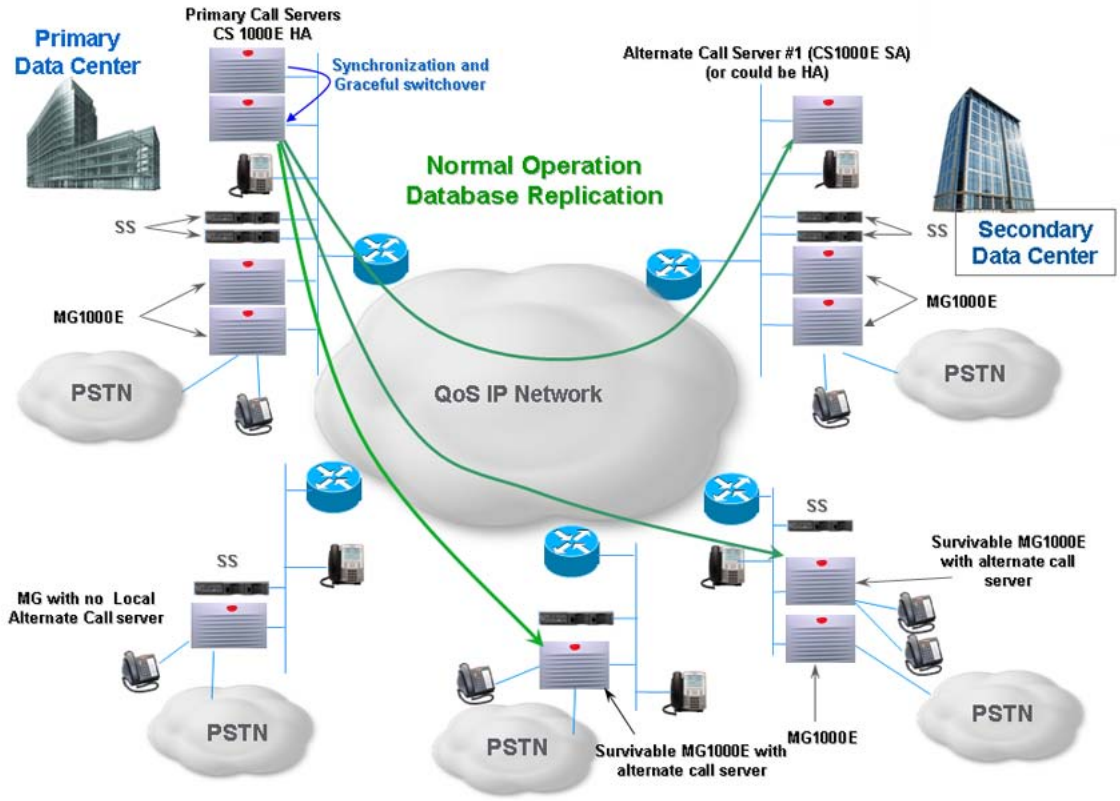

Figure 3: Survivable Media Gateway: Database Replication

Figure 4: Survivable Media Gateway: Primary Call Server Failure or Loss of Network Connectivity on page 29 shows how the system reacts if the Primary Call Server fails or experiences a loss of network connectivity. The Media Gateways register to the Secondary Call Server configured as Alternate Call Server 1.

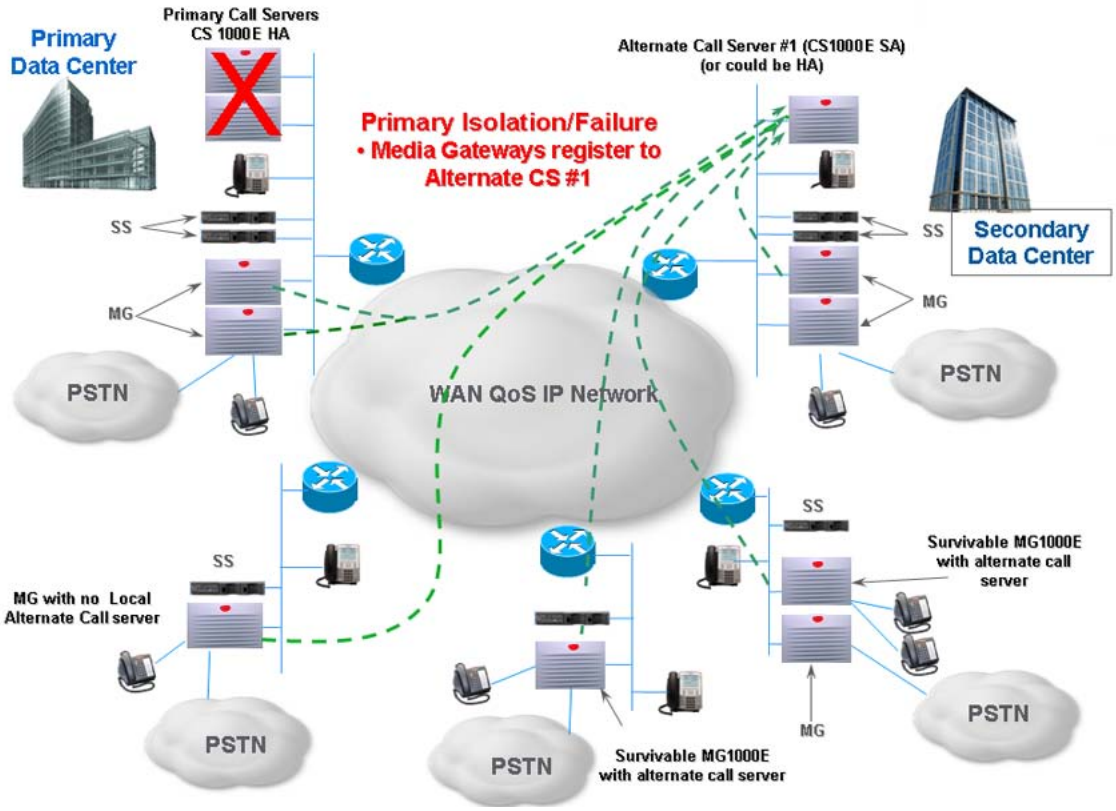

Figure 4: Survivable Media Gateway: Primary Call Server Failure or Loss of Network Connectivity

Figure 5: Survivable Media Gateway: WAN Failure on page 30 shows how the system reacts in the event of WAN failure. The Survivable Media Gateways register to the Secondary Call Server configured as Alternate Call Server 2, typically the Call Server in the local area of the user. Non-survivable Media Gateways are inaccessible due to WAN failure.

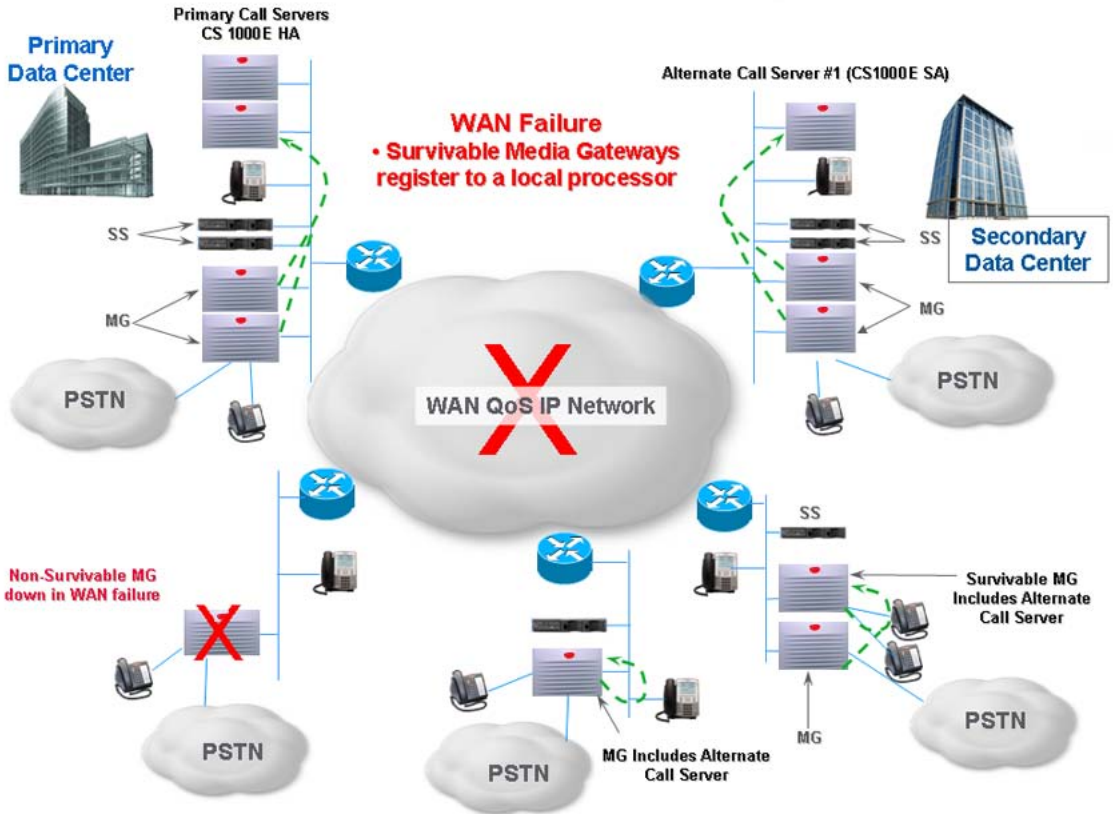

Figure 5: Survivable Media Gateway: WAN Failure

For Survivable SIP Media Gateways case figures 2, 3, 4, and 5 can be re-worked to have one site Survivable SIP Media Gateways. In this case for figure 2, 4 and 5 the MGC in the Survivable SIP Media Gateway side should be registered in normal/survivable modes to the SIP GW. For figure 3, the SIP GW component will not have database replication from the PCS.

A Survivable SIP Media Gateway is similar to the traditional Survivable Media Gateway (SMG) for which the Survivable Call Server receives the database from the Primary Call Server and all IP Phones are redirected to the Primary Call Server during normal operations. However, unlike the SMG, an additional blade called a SIP Media Gateway Controller is used. The Media Gateway Card is always registered to the SIP Media Gateway Controller in normal and survival modes. All TDM devices, such as telephones and trunks, are configured on the SIP Media Gateway Controller.

This architecture separates the IP resources from the traditional digital resources by having two separate call servers:

- One call server (and the associated signaling proxies), known as the Survivable Server, handles the survival aspect of GR which includes receiving the database from the Primary Call Server, providing registration and access services to IP endpoints, for example, IP Phones.
- The second call server (and the associated signaling proxies), known as the SIP Media Gateway Controller, handles all the digital resources at the site, such as traditional TDM and analog phones, fax machines, and PRI/BRI trunk connectivity. These two systems

are usually collocated in the same chassis. The choice of hardware is dictated by the number of users at the location which the Survival SIP Media Gateway servers.

The Media Gateway Controller card and the associated DSPs register to the SIP Media Gateway Controller. The SIP Media Gateway Controller connects the system to the digital resources through the SIP signaling interface.

The Survivable SIP Media Gateway architecture is intended to remove the 80 ms Round Trip Delay (RTD) restriction imposed on the GR network because the Media Gateway Controller in the Media Gateway now registers to the local SIP Media Gateway Controller and no longer has to traverse the network to register to the Primary Call Server. TN space is also freed up on the Primary Call Server because the data on the SIP Media Gateway Controller (virtual loops and TNs for the Media Gateway Controller resources and the associated digital/ analog line devices) does not need to be configured on the Primary Call Server. This allows the Primary Call Server to support up to 511 survivable branches equipped with Survivable SIP Media Gateway.

The SIP Gateway portion of the Survivable SIP Media Gateway provides additional backup to the MAS servers in the network for ringback tone generation in the case of a WAN outage. Migration details from SMG to Survivable SIP Media Gateway can be found in Hardware Migration on page 171.

Software

Survivable Media Gateway is supported only in Communication Server Release 5.0 and later. This software must be installed on the Primary Call Server and all Secondary Call Servers in the system. In addition, Package 404 (GRPRIM) must be installed on the Primary system and Package 405 (GRSEC) must be installed on the Secondary systems.

Hardware

Survivable Media Gateway uses triple IP registration. For more information about triple IP registration and compatible hardware devices, see Triple IP registration on page 33.

Database replication

Important:

In a Survivable Media Gateway configuration, the database is configured and modified on the Primary Call Server and replicated manually or automatically to all Secondary Call Servers.

Starting with Communication Server 1000 Release 6.0, database replication occurs by Secure File Transfer Protocol (SFTP). In Survivable Media Gateway configurations where some of the Call Servers have not yet been migrated to Release 6.0 or later software,

database replication occurs using regular FTP until all Call Servers in the configuration have been migrated to Release 6.0 or later software.

During each database replication, the Primary Call Server performs the following steps:

- creates a copy of the database backup files to a temporary directory on its hard drive
- uses TAR to group the directory into one file
- uses GZIP to compress the file
- transfers the compressed file to a specific directory on the hard drive of each Secondary Call Server configured in the system

To complete the replication process, each Secondary Call Server performs the following steps:

- restores the database received from the Primary Call Server
- performs a sysload to endorse the received database

Note:

System-specific data, such as IP addresses, netmasks, routes, and nodes, is included in the replicated database but is filtered out and not restored on the Secondary Call Servers.

For more information about manual database replication, see Manual database replication and restore on page 85. For more information about scheduled database replication, see Configure Database Replication Control (GRDRC) block on page 65.

Databases are replicated from the Primary Call Server to all configured Secondary Call Servers using a defined schedule. If replications are scheduled to occur at the same time, the transfers occur sequentially, one at a time. The actual transfer time depends on the size of the database and the ELAN bandwidth between the sites. An approximation is 5 minutes for each replication.

The database replication includes invoking a database dump at the start of each schedule implementation. This is done to ensure that any previous changes are saved. If the schedules are set to occur at the same time then only one database dump is performed. This can reduce the overall database replication activity if a large number of remote call servers are being used in a Geo-Redundancy configuration.

To increase the security of database replication, the zipped database is encrypted on the Primary Call Server before it is replicated to the Secondary Call Servers. The zipped file is then decrypted on each Secondary Call Server. If the decryption fails, the database is not restored.

Normal operation

A Geographic Redundancy configuration using Survivable Media Gateway consists of the following components:

- one Primary Call Server
- up to 50 Secondary Call Servers
- up to 50 Media Gateways controlled by Media Gateway Controllers

Each Secondary Call Server in the system can be configured as Alternate Call Server 1 (AC1) or Alternate Call Server 2 (AC2) for all or part of the system resources.

When Survivable SIP Media Gateway components are used, 511 survivable Call Servers are supported for each PCS.

Note:

Avaya recommends that the Secondary Call Server that is configured as Alternate Call Server 2 be in the same local area as its assigned resources. This helps to provide redundancy in the event of WAN failure.

You can implement Survivable Media Gateway using different system types. The Primary and Secondary Call Servers can be any combination of Avaya CS 1000E (CP PIV, CP PM) systems.

Triple IP registration

Triple IP registration is no longer required in the Survivable SIP Media Gateways architecture. The Media Gateway Controllers and all other media cards in the Media Gateway component register locally to their own Call Server. Triple IP registration works the same as Release 6.0 for Media Gateways operating in the SMG configuration. It establishes survivability over layer 3 network connections for TDM resources. Therefore, you can configure a device with first, second, and third choice preferences for Call Server IP addresses.

The following devices support triple IP registration:

- Media Gateway 1000E (with Media Gateway Controller)
- ITG-SA
- Media Card 32S

Note:

ITG-SA and Media Card 32S support triple IP registration for voice Media Gateway applications only. ITG-SA and Media Card 32S must be in the same ELAN subnet as the Signaling Server leader they receive their initial bootload from. If the Media Gateway housing these cards is to be remotely deployed over a layer 3 network, there must be a signaling

server leader installed at the same location on the same local ELAN subnet. The Media Gateway Controller (MGC) and its hosted MGC DSP daughterboards use a different boot mechanism and do not have the same ELAN subnet restriction.

Under normal operating conditions, all devices are registered to the Primary Call Server. In Survivable SIP Media Gateways architecture TDM resources are registered to the SIP GW component. If the Primary Call Server fails, the device registers to the assigned Alternate Call Server 1 (see Figure 4: Survivable Media Gateway: Primary Call Server Failure or Loss of Network Connectivity on page 29). If the WAN fails, the device registers to the assigned Alternate Call Server 2 (see Figure 5: Survivable Media Gateway: WAN Failure on page 30).

In a Survivable Media Gateway component a Media Gateway that loses connection to its Primary Call Server may first attempt to connect to the Primary Call Server using an alternative connection as provided by the Media Gateway Dual Homing feature. If the dual-homing connection attempt fails, the Media Gateway will then attempt to register to the Alternate Call Server 1. If a Media Gateway cannot connect to the Alternate Call Server 1, it registers with the assigned Alternate Call Server 2.

Important:

The Media Gateway Controller is registered to the same Call Server as the Media Cards associated with the MGC. In a Survivable SIP Media Gateway component Media Gateways are registered locally with SIP GW component and connected to the PCS via virtual trunks.

When a Media Gateway 1000E (MG 1000E) registers to a Call Server, the Call Server sends a message to all the Media Cards on that MG 1000E to register to it.

For Survivable Media Gateway components ITG-SA, Media Card 32S, and MGC DSP Daughterboards are configured on the Primary Call Server. The Primary Call Server replicates these settings to the Secondary Call Servers during database replication.

Media Gateway Controllers and Media Cards are configured on the Primary Call Server using Element Manager or through the command line interface of the Media Gateway Controller. Although it is physically possible to use Element Manager to directly access the Secondary Call Servers and make changes to the MGC configuration, this should not be done, as the data will be overwritten the next time the Primary Call Server does a database replication to the Secondary Call Server. Figure 20: Element Manager IPMG 40 Property Configuration Web page on page 75 shows an example of Media Gateway Controller configuration using Element Manager.

For more information about configuring the Media Gateway Controller, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

Vacant Number Routing for TDM digital and analog telephones

Vacant Number Routing (VNR) for TDM digital and analog telephones is supported. The TDM telephone is defined on a Media Gateway registered on one of the Secondary Call Servers in

the Survivable Media Gateway configuration. TDM vacant numbers are routed to the NRS, which resolves the location of the Media Gateway where the TDM telephone is defined. VNR limitations for IP calls also apply to TDM calls.

No VNR specific configuration activity is required for Geographic Redundancy to work. However, VNR capability can be used in some configuration scenarios to enhance call connections in failover situations.

For more information about how VNR and NRS handle calls during Primary Call Server failure or network connectivity issues, see Network connectivity failure call scenarios on page 40.

Active Call Failover

Active Call Failover (ACF) ensures that active calls are not dropped during switchover. If Geographic Redundancy IP Phone failover occurs while an IP Phone is in an active call, the call can stay active until the user hangs up. No call features (like Transfer or Conference) can occur in ACF state. Once the user hangs up, the IP Phone will immediately redirect. ACF is only valid for IP Phone to IP Phone calls. If the IP user was calling through a Media Gateway (trunk call or TDM or analog phone) that also fails over, the phone call will be disconnected as all Media Gateway resources reset when the Media Gateway fails over.

For more information about Active Call Failover, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

Abnormal operation

Primary system failure

With Geographic Redundancy, the transfer of system control from a failed Primary system to a Secondary system is not the same as a traditional graceful switchover operation. There will be a period of time that IP Phones and Media Gateway resources will not provide service. For a Survivable Media Gateway (SMG), there is a time-out period (default 120 seconds) where the SMG will wait to see if the Primary system becomes active before they failover to an Alternate Call Server. This allows a SMG to tolerate a Primary Call Server restart or a temporary WAN outage (such as a router being reset).

When a Survivable Media Gateway does failover, it also resets, which can take up to two minutes before service is restored. During this reset, all TDM resources in the Media Gateway will also be reset, so all active calls will be lost.

For IP Phones, if the Secondary system cannot redirect the IP Phones to the Primary system because of a Primary system failure or network connectivity problem, the IP Phones remain registered on the Secondary system.

If a Primary Call Server fails or becomes unreachable, the IP Phone will time-out, reset, and attempt to register again to its S1, where it receives local survivable service from the Secondary Call Server. When the Primary Call Server is back online, the Secondary Call Server redirects the IP Phone back to the Primary Call Server.

The time it takes for all IP Phones to failover or recover depends on the number of IP Phones registered to each signaling server and the total number of IP Phones trying to re-register to any particular Call Server. In a large system with 5000 IP Phones registered to a signaling server, it could take up to 20 minutes for the last IP Phone to re-register and regain service.

The failure scenario and subsequent re-registration of IP Phones to the local survival Call Servers remains the same in case of Survivable SIP Media Gateway component. The timeout period of 120 sec for the SMG to failover to the Survival Call Server and become active is valid for the SCS component of the Survivable SIP Media Gateway as well. When the survival Call Server resets during failover, the Media Gateway component or the associated TDM resources of the Survivable SIP Media Gateway does not reset. The TDM resources (the Media Gateway Controller) remain registered to the SIP gateway component of the Survivable SIP Media Gateway because this component is not involved in geographic redundancy operation or the active failover processes. The other aspects of failover, like secondary system operating states, secondary system failure, network connectivity failure call scenarios, remain unchanged.

Note:

Failure of the IP Phone to register to the Primary system is not necessarily caused by a failure of the Primary system. The cause can also be LAN/WAN connectivity problems. Geographic Redundancy does not differentiate between these types of failures.

To monitor Primary system health, the Secondary system maintains a real-time count of IP Phones (N1) and Media Gateways (N2) registered on the Secondary system. When the number of IP Phones (N1) or Media Gateways (N2) registered on the Secondary system exceeds the Geographic Redundancy thresholds for IP Phones (GRTHR1) and Media Gateways (GRTHR2), the Secondary system escalates to ACTIVATING state. (GRTHR1 and GRTHR2 are defined in LD 117.)

Secondary system operating states

The Secondary system provides a number of operating states that allow smooth transition from INACTIVE to ACTIVE and back again. See Figure 6: Secondary system operating state logic on page 37 and Table 2: Secondary system state control data set on page 38 for a description of the Secondary system state logic and the data set, including timers, that control the transitions between the various Secondary system states.

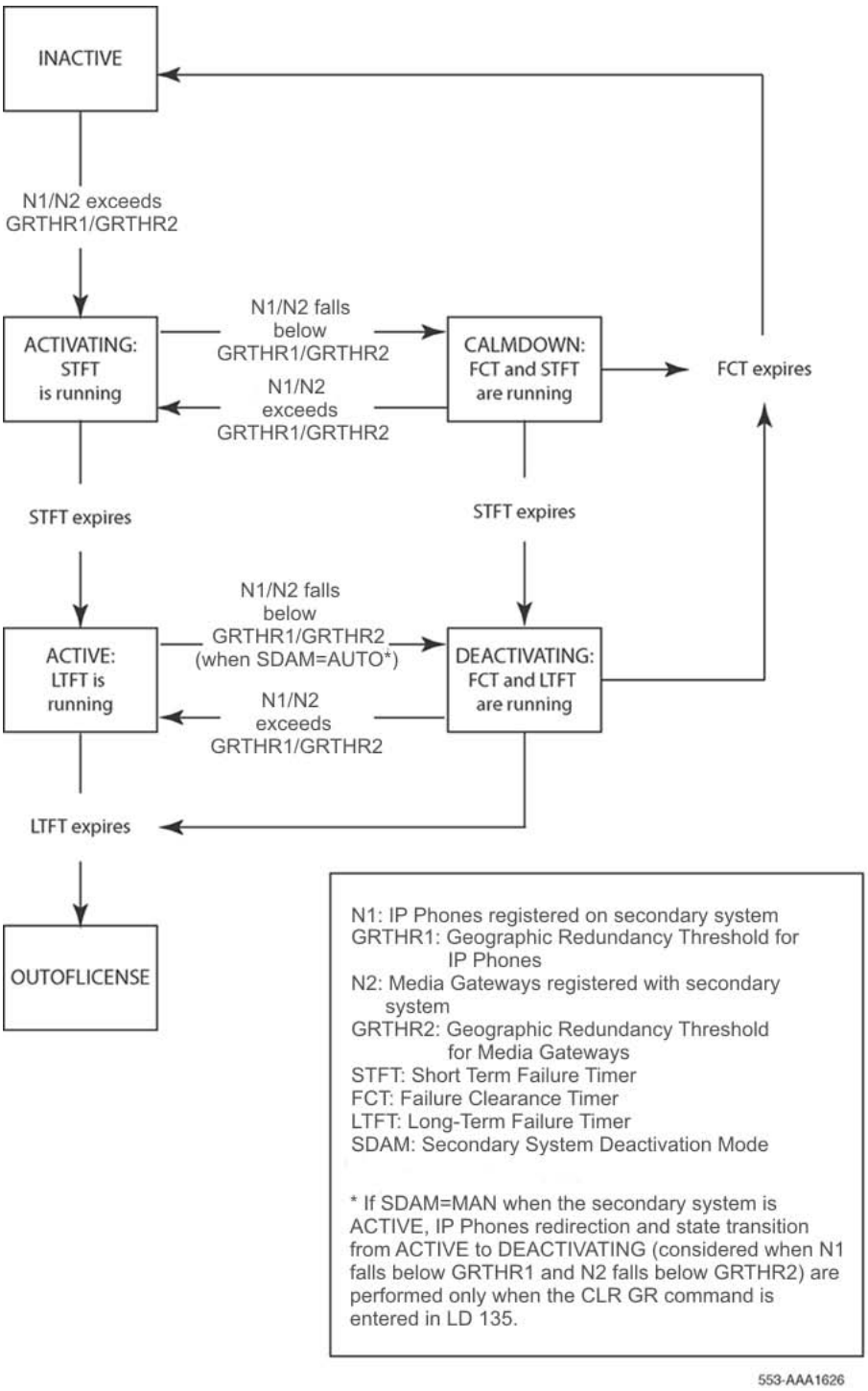

Figure 6: Secondary system operating state logic

Table 2: Secondary system state control data set

Variable	Definition
N1	Number of IP Phones registered with the Secondary system, updated in real time. The Secondary system uses this value to monitor Primary system health.
N2	Number of Media Gateways registered with the Secondary system, updated in real time. The Secondary system uses this value to monitor Primary system health.
GRTHR1	Geographic Redundancy threshold for IP Phones. This is a customer-definable threshold against which N1 is compared. If the Secondary system is INACTIVE and N1 exceeds the GRTHR1, the Secondary system escalates to the ACTIVATING state. (GRTHR1 is configurable in LD 117.)
GRTHR2	Geographic Redundancy threshold for Media Gateways. This is a customer-definable threshold against which N2 is compared. If the Secondary system is INACTIVE and N2 exceeds the GRTHR2, the Secondary system escalates to the ACTIVATING state. (GRTHR2 is configurable in LD 117.)
STFT	Short Term Failure Timer: the short period of time during which the Primary system can remain inoperable without the Secondary system assuming full system control (for example, as caused by initialization, sysload, or minor technical failure). STFT starts when the Secondary system enters the ACTIVATING state (N exceeds GRTHR). If the STFT expires while the system is in the ACTIVATING state, the Secondary system escalates to the ACTIVE state. (STFT is configurable in LD 117.)
FCT	Failure Clearance Timer: period of time that must elapse when the Primary system is brought back online (and N falls below GRTHR) before the Secondary system state can revert to INACTIVE. This timer prevents unnecessary transitions between less severe and more severe states when the value of N is close to GRTHR. (FCT is configurable in LD 117.) **Note:** FCT is also used with the CLR GR command in LD 135. FCT defines the period of time during which redirection attempts to the Primary system are allowed.
LTFT	Long Term Failure Timer: license period during which the Secondary system is allowed to run in the ACTIVE state. (LTFT is a hardcoded value of 90 days. It is not a customer-configurable value.) When LTFT expires, the system enters the OUTOFLICENSE state.
SDAM	Secondary System Deactivation Mode: specifies whether the Secondary system is in automatic deactivation (AUTO) or manual deactivation (MAN) mode. If SDAM = AUTO, the Secondary system, when ACTIVE, queries the NRS for the Primary system address every ten minutes. When the Primary system comes back online the Secondary system can resume the redirection of IP

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Variable	Definition
	Phones to the Primary system and automatically revert to the DEACTIVATING state (when N1 falls below GRTHR1 and N2 is below GRTHR2). If SDAM = MAN, the Secondary system, when ACTIVE, stops trying to redirect IP Phones to the Primary system. It stays in the ACTIVE state until the CLR GR command is initiated manually in LD 135. The CLR GR command triggers the redirection attempts to the Primary system to resume (for a maximum period defined by FCT). If the redirections are successful and N1 falls below GRTHR1 and N2 falls below GRTHR2, the Secondary system reverts to the DEACTIVATING state. (SDAM is configurable in LD 117.)

For details on configuring the Geographic Redundancy State Control Block, refer to Configure State Control (GRSC) Blocks on page 70.

For details on the recovery of the Primary system following a Long-Term Failure, refer to Recovering the Primary system database on page 82.

Secondary system operating state survival

The Secondary system operating state can survive an initialization or a sysload (the operating state is restored from non-volatile memory). However, if the operating state is restored to ACTIVE after a sysload or initialization, the state transition from ACTIVE to DEACTIVATING is prevented for 15 minutes. This waiting period eliminates unnecessary state transitions so that IP Phones can re-register to the Secondary system. The Secondary system can now obtain the appropriate value of IP Phones registered (N1).

OUTOFLICENSE state

When the Secondary system remains in the active state beyond the Long Term Failure Timer (LTFT) of 90 days and enters the OUTOFLICENSE state, some functional limitations are imposed on the system. `Beyond licensed period` appears on all IP Phones and `licensed period is exceeded` appears on the TTY banner upon successful logon.

As well, all the data dump (midnight and manual), automatic restore, and automatic sysload operations are restricted.

To clear the OUTOFLICENSE state, a software installation is required.

Secondary system failure

If the Primary system is active, TDM resources and IP Phones receive services from the Primary system.

If the Primary Call Server fails, the TDM resources and IP Phones receive services from Alternate Call Server 1.

If both the Primary and Alternate Call Server 1 Call Servers fail, the TDM resources and IP Phones receive services from Alternate Call Server 2.

If Alternate Call Server 1 is down but the Primary Call Server and Alternate Call Server 2 are active, the TDM resources and IP Phones receive services from the Primary system.

If Alternate Call Server 2 is down but the Primary system is active, the TDM resources and IP Phones receive services from the Primary system.

If the Primary and Alternate Call Server 2 systems are down, the IP Phones may not be able to get service as neither the S1 or S2 signaling servers is accessible.

When its S1 Retry Count expires, the IP Phone uses its S2 value to register directly to the Primary system and receive service. TDM resources continue to provide service according to the triple IP registration settings. If connectivity to the Primary server is lost while the Secondary Call Server is in a failed state, the resources attempt to connect to the Secondary Call Server defined as Alternate Call Server 2.

Network connectivity failure call scenarios

Table 3: VNR and NRS routing when Primary Call Server fails on page 40 and Table 4: VNR and NRS routing when Primary Call Server is active but unreachable on page 41 show how NRS and VNR handle various calls when connectivity has failed due to Primary Call Server failure and network issues.

When the Primary Call Server fails, VNR and NRS respond as shown in Table 3: VNR and NRS routing when Primary Call Server fails on page 40

Note:

No VNR specific configuration activity is required for Geographic Redundancy to work. However, VNR capability can be used in some configuration scenarios to enhance call connections in failover situations.

Table 3: VNR and NRS routing when Primary Call Server fails

	Call originator	Call destination	Call treatment
Internal calls	IP Phone registered on a Secondary CS in the system	IP Phone registered on a different Secondary CS in the system	VNR routes incoming calls to a Secondary CS (Alternate Call Server 1 or Alternate Call Server 2)
	TDM telephone configured on an MG registered to a Secondary CS in the system in case of SMG component. If	TDM telephone configured on an MG registered to a different Secondary CS in the system in case of SMG	

	Call originator	Call destination	Call treatment
	this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component, no reboot.	component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component, no reboot.	
Incoming external calls	Any telephone on a non-Geographic Redundant system	IP Phone registered on a Secondary CS in the system	NRS routes incoming external calls to a Secondary CS (Alternate Call Server 1 or Alternate Call Server 2)
		TDM telephone configured on an MG registered to a Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component, no reboot.	

If connectivity problems allow some Secondary Call Servers to communicate with the Primary system while others cannot, VNR and NRS route the calls as shown in Table 4: VNR and NRS routing when Primary Call Server is active but unreachable on page 41.

Table 4: VNR and NRS routing when Primary Call Server is active but unreachable

	Call originator	Call destination	Call treatment
Internal calls	IP Phone registered on active Secondary CS in the system	IP Phone registered on Primary CS	VNR routes calls to a Secondary CS (Alternate Call Server 1 or Alternate Call Server 2)
	TDM telephone configured on an MG registered to active Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	TDM telephone configured on an MG registered to Primary CS in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	

	Call originator	Call destination	Call treatment
	IP Phone registered on active Secondary CS in the system	IP Phone registered on a different Secondary CS in the system	NRS routes calls to Primary CS, which responds according to HUNT or CFNA settings
	TDM telephone configured on an MG registered to active Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	TDM telephone configured on an MG registered to a different Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	
	IP Phone registered on Primary CS	IP Phone registered on active Secondary CS in the system	NRS routes calls to Primary CS, which responds according to HUNT or CFNA settings
	TDM telephone configured on an MG registered to Primary CS in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	TDM telephone configured on an MG registered to active Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	
Incoming external calls	Any telephone configured on a non-Geographic Redundant system	IP Phone registered on Primary CS	NRS routes call to Primary CS
		TDM telephone configured on an MG registered to Primary CS in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	

	Call originator	Call destination	Call treatment
	Any telephone configured on a non-Geographic Redundant system	IP Phone registered on active Secondary CS in the system	NRS routes calls to Primary CS, which responds according to HUNT or CFNA settings
		TDM telephone configured on an MG registered to active Secondary CS in the system in case of SMG component. If this is Survivable SIP Media Gateway configuration, TDM resources are registered with SIP GW component.	

Survivable Media Gateway planning

For a list of Call Servers supporting Survivable Media Gateway configuration, refer to Normal operation on page 33. Plan the Primary system according to the standard Avaya CS 1000E configuration steps, add the Secondary Call Servers at the desired survivable locations, and then adjust the configuration of the local Media Gateways to accommodate the Secondary Call Servers.

For more information about Avaya CS 1000E planning considerations, see *Avaya Communication Server 1000E Planning and Engineering, NN43041-220*.

Common Avaya CS 1000E planning considerations

In a Survivable Media Gateway configuration, the Secondary systems do not have to be duplicates of the Primary system. Each Secondary Call Server does have an exact copy of the Primary system database, so it must be configured the same, but it might only be supporting a subset of the phones and resources of the Primary Call Server. Depending on how it is deployed, a Secondary Call Server could be configured to service all IP Phones and Media Gateways or just a small subset. In a geographically dispersed system, the Secondary Call Servers may be logically grouped to support IP Phones and Media Gateways in the same location or region. WAN topology must be considered when engineering these survivable clusters.

The following are considerations to make when planning a Survivable Media Gateway configuration.

Signaling Servers

The Secondary Avaya CS 1000E systems, like the Primary system, must have Signaling Servers installed to provide service to IP Phones. Configure Signaling Servers at the secondary site independently from Signaling Servers at the Primary site. Configuration settings for Signaling Servers are not included with database replication so they must be configured individually.

The number and configuration of Signaling Servers in the Primary and Secondary systems do not need to be the same. If traffic and capacity requirements are lower when the Secondary system becomes active, install fewer Signaling Servers at the Secondary site. When this is the case, you can install enough Signaling Servers to handle the traffic created when the Secondary system becomes active.

For more information about Signaling Server capacity, see *Avaya Communication Server 1000E Planning and Engineering, NN43041-220* and *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

Co-resident Call Processor and Signaling Server

CS 1000 Release 6.0 introduces the Co-resident Call Processor and Signaling Server, which is capable of running the Call Server software, Signaling Server software, and System Management software on the same hardware platform operating under the RedHat Linux Operating System. For CS 1000 Release 6.0, the only supported hardware platform for the Co-resident Call Processor and Signaling Server is the Call Processor-Pentium Mobile (CP PM) platform.

Note:

In CS 1000 Release 6.0, the Co-resident Call Processor and Signaling Server does not support an HA configuration (dual core with Active/Inactive role). For systems that require HA configuration, the VxWorks-based Call Server software must be deployed.

In a Co-resident Call Processor and Signaling Server, there is a default route on the TLAN interface which causes an error message to occur after initiating file transfer requests between redundant systems (for example, during backups). To ensure normal system operation in redundant Co-resident Call Processor and Signaling Servers, you must create routes to the remote ELANs of the other Call Servers using the ELAN gateway to force this traffic to use the ELAN as the source IP address. You can create these routes using the Linux base manager.

For more information about Co-resident Call Processor and Signaling Server, see *Avaya CP PM Co-resident Call Server and Signaling Server Fundamentals, NN43001-509*.

Comments? infodev@avaya.com

Voice Gateway channels

The Voice Gateway channels are also called DSP channels or the card's physical TNs. The Voice Gateway channels are supported by the DSP daughterboard on the Media Gateway Controller, the ITG-SA card, and the Media Card 32S.

For more information about configuring Voice Gateway channels, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

NRS

To support the IP Phone redirection process and to ensure that each system can properly route calls following a system failure, a Primary and Alternate Network Redirect Server (NRS) are required on the network. The Alternate NRS periodically synchronizes its database with the Primary NRS. This ensures that if the Primary NRS fails, the Alternate NRS can assume the role of the Primary NRS.

For more information on NRS, see NRS routing for IP Phone redirection on page 103.

To provide the necessary redundancy, install each NRS in a different location. Install one NRS with each system or install each NRS in a remote location apart from either system. Wherever they are installed, ensure that at least one NRS remains operational following failure of either system.

For more information on installing and configuring the Primary and Alternate NRS, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

NCS

Network Connect Server (NCS) is an application associated with the NRS that supports Geographic Redundancy. The NCS allows the Secondary system node TPS to query the NRS directly to perform the redirection of IP Phones to the Primary system.

You must configure NCS properties when the Primary and Secondary endpoints are defined on the NRS and when the Primary and Secondary system IP telephony nodes are defined in Element Manager.

For more information, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*, *Avaya Network Routing Service Fundamentals, NN43001-130*, and NRS routing for IP Phone redirection on page 103.

Network Time Protocol

Network Time Protocol (NTP) is used to synchronize computer clocks in the global IP network. NTP provides a comprehensive mechanism to access the national time and frequency

dissemination services, organize the time-synchronization subnet, and adjust the local clock in each participating subnet peer.

Because the parameters for NTP depend upon the geographic location, the NTP parameters stored on the Primary Call Server cannot be replicated to the Secondary Call Server as it is in a different geographic location. The NTP parameters are stored in a separate database file that is not replicated to the Secondary Call Server. Therefore, an NTP configuration does not survive a geographic redundancy switchover.

Note:

In Co-resident systems, NTP is managed by the Linux operating system and configured using Linux Base commands or Element Manager. The LD 117 overlay commands are not available.

For information about Network Time Protocol, see *Avaya Features and Services Fundamentals, NN43001-106*.

Time of Day clock

The Secondary system must always have the same time of day local clock as the Primary system, regardless of the actual time zone of the secondary site.

Note:

In Co-resident systems, Time of Day (TOD) is managed by the Linux operating system and configured using Linux Base commands or Element Manager. Overlay commands are not available.

Network considerations

The Primary and Secondary systems must comply with network requirements as described in *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*. In addition, install the Primary and Secondary system TLANs in different LAN/WAN subnets; however, you can install both systems in the same ELAN subnets. The systems must use the same NRS for routing.

The maximum round-trip delay on the IP network between Call Servers and Media Gateways is 80ms, with a maximum allowable packet loss of 0.5% (0% is recommended).

Firewalls

The database-replication process requires secure FTP access through the WAN between the Primary and Secondary systems. Therefore, the proper TCP/IP ports must be open on the appropriate security firewall servers.

Comments? infodev@avaya.com

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

ISSS/IPsec considerations

All Secondary Call Servers must be configured with the same security level and pre-shared key (PSK) as that of the Primary Call Server.

For more information about ISSS/IPsec, see *Avaya Security Management Fundamentals, NN43001-604*.

UNIStim with DTLS signaling encryption

There are certain limitations to consider when enabling UNIStim DTLS (Datagram Transport Layer Security) signaling encryption in systems configured for Geographic Redundancy. If an IP Phone is configured with action byte 7 (Secure Handshake Registration Mode), you must ensure that every UNIStim server the phone can register to supports DTLS and is properly configured. The Primary and Secondary systems must both support DTLS.

Geographic Redundancy functions normally if the Basic security setting for UNIStim DTLS signaling encryption is configured. However, if the systems and hardware do not meet the DTLS requirements, enabling UNIStim security with Advanced or Complete settings may cause Geographic Redundancy to fail.

For more information about UNIStim with DTLS signaling encryption, see *Avaya Security Management Fundamentals, NN43001-604*.

SIP Line

The SIP Line Service fully integrates Session Initiation Protocol (SIP) endpoints in the CS 1000 system and extends the CS 1000 telephony features to the SIP clients.

For information about SIP Line and Geographic Redundancy, see *Avaya SIP Line Fundamentals, NN43001-508*.

Unicode Name Directory

Unicode Name Directory (UND) provides line ID localized name lookup on the Primary Call Server only. When service switches over to a redundant Secondary Call Server, the UND service is not available.

For information about Unicode Name Directory, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

Zone based resource allocation

The Call Server algorithm to select the majority of trunking resources is based on the "trunking gateway" paradigm, where the Avaya CS 1000E routes the call to a specific trunk location based on a dialling prefix. This remains in effect for COT, DID (and DID-like trunks, such as FEX, WATS, and so on) and TIE trunks.

This algorithm does not include the selection of trunk resources from the same zone as the caller. It provides a destination based route selection, as opposed to an originator based route selection. If it is required to route calls over trunks in the same zone as the caller, then the dialed number needs to be prefixed in a manner allowing the pre-translated number to uniquely match a calling zone to the trunk routes pertaining to that zone.

However, the algorithm for Music, RAN, Conference, Tone and Digit Switches, and a limited subset of Local trunks does search for resources in the same zone as the user. If the appropriate resource has been configured in the local zone and is idle, that resource will be selected first. If the resource is not configured locally or is busy, other system-wide resources are used. This order of resource allocation can conserve WAN bandwidth as Media Gateways in a specific geographic location typically share the same zone number. The local resource is useful in saving WAN bandwidth between sites.

For more details about the potential bandwidth that could be conserved, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*, which defines bandwidth requirements by call and by codec type.

Survivable Media Gateway configuration

Before configuring a system for Survivable Media Gateway, an Avaya CS 1000E must already be installed as the Primary system.

For information about installing and configuring an Avaya CS 1000E Call Server, see *Avaya Communication Server 1000E Installation and Commissionin, NN43041-310*. For information about upgrading an Avaya CS 1000E system, see *Avaya Communication Server 1000E Upgrades, NN43041-458*.

Two software packages support the Survivable Media Gateway configuration: package 404 (GRPRIM) installed on the Primary system and package 405 (GRSEC) installed on all Secondary systems. These packages are mutually exclusive and allow the customer database on the Primary system to be regularly replicated to the Secondary systems across the Wide Area Network (WAN).

Note:

The GRSEC package (405) must be equipped during the software installation process; you cannot add it using a new keycode after the software has been installed.

Comments? infodev@avaya.com

Use the following procedure to configure Survivable Media Gateway on the Primary Call Server.

Configuring a Survivable Media Gateway system

1. Install the Primary Call Server (PCS).
   - Install hardware (if required).
   - Install Avaya CS 1000 Release 6.0 software, including GRPRIM package.
   - Configure IPMGs and Signaling Servers.
   - Configure NRS.

     (Also see NRS routing for IP Phone redirection on page 103for more information on NRS and Geographic Redundancy.)
   - Install the customer database (if required).

   For information about installing and configuring an Avaya CS 1000E system, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.
2. Configure the Primary system for Survivable Media Gateway. For more information, see Configuring Database Replication on the Primary Call Server on page 58.
3. Install Secondary Call Server (SCS) on existing IPMG or a new IPMG:
   - Install hardware (if required). This SCS can be any one of the following:
     - CP MG Call Server (Linux-based CPL)
     - CP PM Call Server (Linux-based CPL)
     - CP PM Call Server. (VxWorks)
     - COTS Call Server (Linux based CPL)
   - Install the software. The keycode must include package 405 (GRSEC).
   - Install the default customer database.

     For information about installing and configuring an Avaya CS 1000E system, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310* .
4. Install and configure the Media Gateway Controllers.

   Note:

   Configure the Media Gateway Controllers on the Primary Call Server. The settings are included with database replication to the Secondary Call Servers.

   Figure 7: Element Manager Media Gateway configuration screen on page 50 shows the Element Manager Media Gateway configuration page.

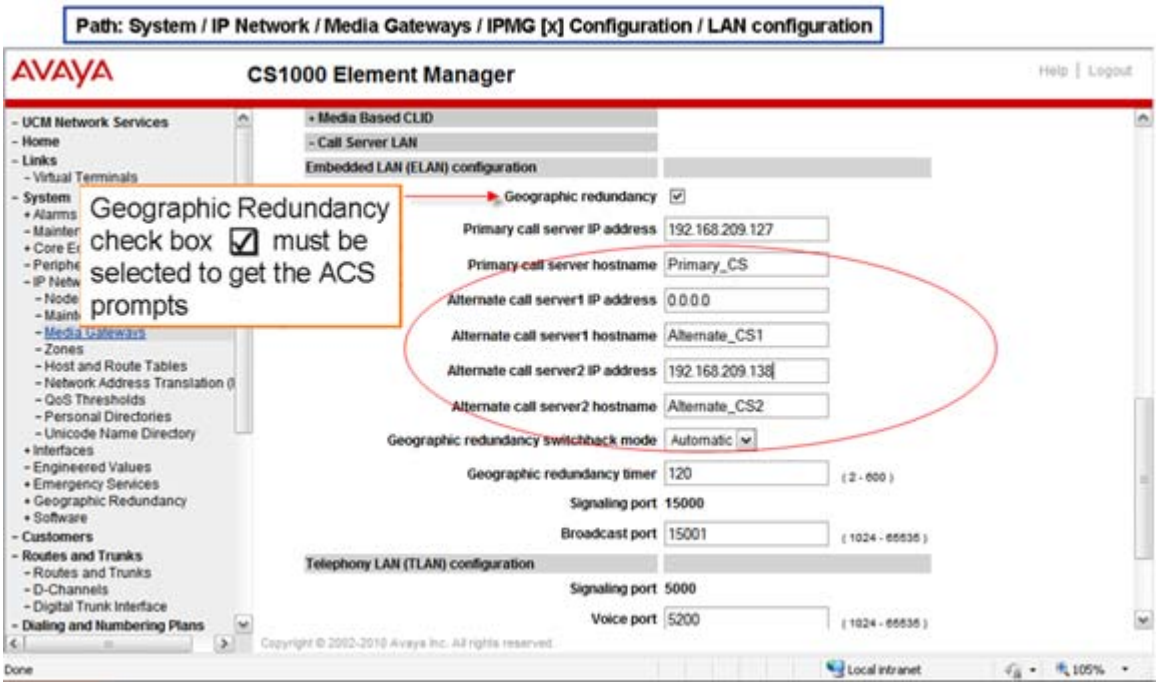

Figure 7: Element Manager Media Gateway configuration screen

5. Install and configure the Signaling Servers. Create a node in the SCS that has the same node number as the PCS.
6. Install and configure the Media Cards.

   To enhance survivability for the larger system, configure the Media Cards on the local servers in the Survivable Media Gateway IP telephony node as a DSP daughterboard or Media Card resource for the node with the appropriate Primary Call Server, Alternate Call Server 1, and Alternate Call Server 2.

   If the Media Cards are in a remote Media Gateway, such as one in a different subnet than the Primary Call Server IP telephony node, the MG 1000 must be a survivable MG 1000 with its own Call Server and Signaling Server so that the Media Cards can obtain their configuration properties from the node leader Signaling Server using the bootp process.

   Figure 8: Element Manager Media Card configuration screen on page 51 shows the Element Manager Media Card configuration page.

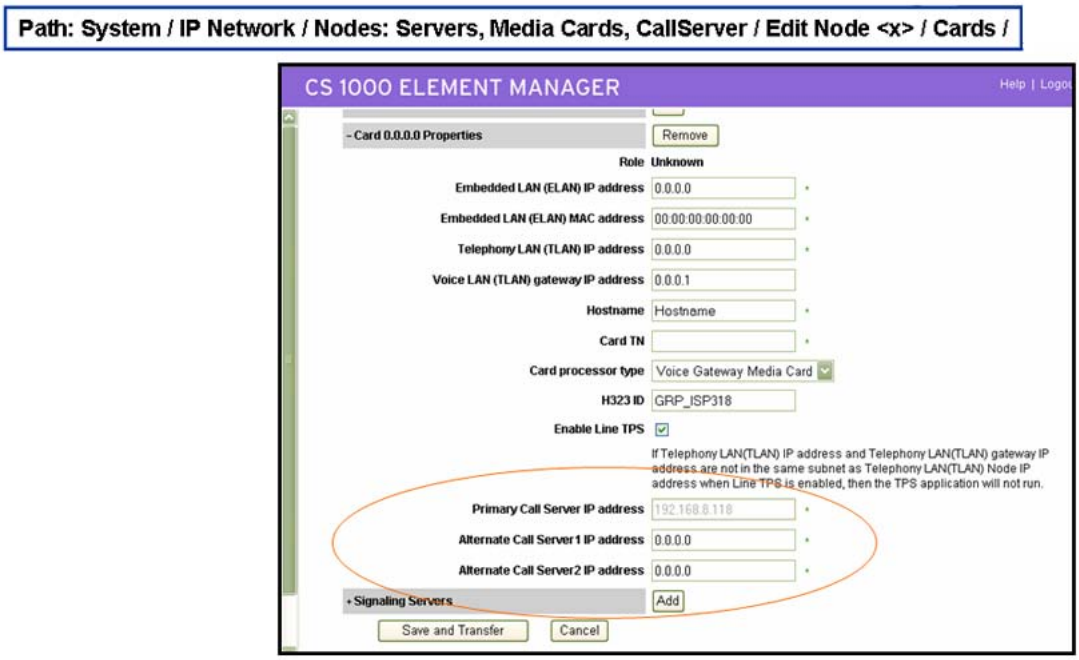

Figure 8: Element Manager Media Card configuration screen

7. Install the Primary and Alternate NRS and configure IP Peer Networking.

   For information about configuring IP Peer Networking, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

8. On the Primary Call Server, transfer the database to the SCS by using the BKR command in LD 43 and specifying the backup rule associated with each Secondary Call Server. For example:

   ```
   .BKR 1
   ```

9. On each Secondary system, perform the following:

   - Log on to the call server and create a temporary backup rule in the default database. For example:

     ```
     new bkpr 1 FMD
     ```

   - Define Backup Rule 1 on the Secondary system for restoring the first database received from the Primary system.
   - Initiate a manual restore from the backup data received from the Primary system by entering the following command in LD 43:

     ```
     RSR 1
     ```

     where 1 is the Restore Rule defined in the GRDRC block (all IP-specific data is removed).

   - Perform a manual sysload in LD 135 by entering the following command:

     ```
     .sysload active
     ```

- Enter `yes` to load the new database.

10. If required, install and configure the IP Phones with S1 pointing to the secondary node TPS and S2 pointing to the Primary system node TPS.

    For more information about installing IP Phones, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

Survivable SIP Media Gateway planning

The list of Call Servers supporting the Survival Call Server portion of the Survivable SIP Media Gateway is the same as the list supported for SMG operation. For the Media Gateway portion or the SIP Gateway, the list of Call Servers supported is expanded to include the CP MG card. Table is in file add1.rtf

		Primary Call Server CPU Type	
		CP PIV	CP PM
Secondary Call Server CPU Type (SCS component of SSMG)	CP PIV		Y
	CP PM		Y
SIP Gateway Call Server CPU Type	CP MG		Y
	CP DC		Y

CS 1000E planning considerations

Each secondary or survival call server does not have to be a duplicate of the primary system. The Secondary Call Server has the exact same database as the Primary Server but it supports only a subset of the phones that the Primary supports.

For a list of Call Servers supporting Survivable Media Gateway configuration, refer to Normal operation on page 33. Plan the Primary system according to the standard Avaya CS 1000E configuration steps, add the Secondary Call Servers at the desired survivable locations, and then adjust the configuration of the local Media Gateways to accommodate the Secondary Call Servers.

For more information about Avaya CS 1000E planning considerations, see *Avaya Communication Server 1000E Planning and Engineering, NN43041-220*.

Signaling Servers

The guidelines for configuring Signaling Servers are the same as Survivable Media Gateways. The Signaling Servers are added to the Secondary Call Server based on traffic and capacity requirements.

Survivable SIP Media Gateways

The Survivable SIP Media Gateway is intended to serve the TDM resources at a location. The planning for Signaling Servers does not need to factor in load considerations for IP phones. The Signaling Server application can co-reside on the CP MG card with the call server application. Additional Signaling Servers may be needed depending on the trunking requirements for the location.

Co-resident Call Processor and Signaling Server

If the Survivable SIP Media Gateway has the Co-resident Call Server and Signaling Server applications deployed to co-reside on the same hardware platform (CP PM, CP MG or CP DC) then the capacity limits for such a deployment need to be considered. The limitations on Co-resident Call Server and Signaling Server applications remain unchanged from Release 6.0.

Survivable SIP Media Gateway/SCS

The CP DC platform can run the Co-resident Call Server and Signaling Server applications. If the location where the Secondary Call Server or Survivable SIP Media Gateway is deployed has more than 1000 IP users, then the CP PM HA VxWorks platform should be used to run the Call Server application.

Survivable SIP Media Gateway SIP G/W

The CMPG platform can run the Co-resident Call Processor and Signaling Server applications. The CP PM hardware along with an MGC card (this configuration is the same as the CS1000E TDM introduced in Release 6.0, with the exception of allowing for SIP Trunking and NRS deployment).

VGW Channels

The VGW channels are configured on the MGC or CP MG card on the Survivable SIP Media Gateway SIP G/W component.

NRS

New concept of the Tertiary NRS is introduced from release 7.0 which is an alternative to the Failsafe NRS in previous releases. The other concepts around the NRS, mainly, deployment, redundancy configurations (Primary/Alternate NRS pair) with database synchronization and routing/proxy operations remain unchanged.

The tertiary NRS provides NRS services when the Survivable SIP Media Gateway or the SMG or any CS1000 SIP GW application cannot contact the Primary or the Alternate NRS. Unlike the failsafe NRS, the tertiary NRS is configured independently and does not receive its database from the Primary NRS. This ensures that the registration data in the tertiary NRS is current and up-to-date.

The tertiary NRS is used in case of a WAN outage at any location in a Geographic Redundancy network and routes call between the SCS and gateway components of the Survivable SIP Media Gateway.

The simplest configuration for an Survivable SIP Media Gateway at a branch or location in a Geographic Redundancy network is to point the Secondary Call Server and the gateway components to each other, ie, in node configuration for the Secondary Call Server put the gateway TLAN IP as the tertiary NRS IP address and vice-versa.

Network Connection Service

The Network Connection Service operation remains unchanged.

Survivable SIP Media Gateway configuration

Before configuring a system for Survivable Media Gateway, an Avaya CS 1000E must already be installed as the Primary system.

For information about installing and configuring an Avaya CS 1000E Call Server, see *Avaya Communication Server 1000E Installation and Commissionin, NN43041-310*. For information about upgrading an Avaya CS 1000E system, see *Avaya Communication Server 1000E Upgrades, NN43041-458*.

Two software packages support the Survivable SIP Media Gateway configuration: package 404 (GRPRIM) installed on the Primary system and package 405 (GRSEC) installed on all Secondary systems. These packages are mutually exclusive and allow the customer database on the Primary system to be regularly replicated to the Secondary systems across the Wide Area Network (WAN).

Note:

The GRSEC package (405) must be equipped during the software installation process; you cannot add it using a new keycode after the software has been installed.

Use the following procedure to configure Survivable Call Server on the Survivable SIP Media Gateway.

Configuring a Survivable Call Server on a Survivable SIP Media Gateway system

1. Install the Primary Call Server (PCS).
   - Install hardware (if required).
   - Install Avaya CS 1000 Release 6.0 software, including GRPRIM package.
   - Configure IPMGs and Signaling Servers.
   - Configure NRS.

     (Also see NRS routing for IP Phone redirection on page 103 for more information on NRS and Geographic Redundancy.)

- Install the customer database (if required).

For information about installing and configuring an Avaya CS 1000E system, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

2. Configure the Primary system for Survivable Media Gateway. For more information, see Configuring Database Replication on the Primary Call Server on page 58.
3. Install Secondary Call Server (SCS) on existing IPMG:
   - Install hardware (if required). This can be a CP PM based Call Server installed within the Media Gateway chassis or a standalone CP PIV based Call Server. The Secondary Call Server can be a Standard Availability (SA) single core or a Redundant High Availability (HA) dual core configuration.
   - Install the software, including GRSEC package. The keycode must include package 405 (GRSEC).
   - Configure the Secondary system-specific customer database.
   - Configure Ethernet Protocol in LD 117 -
   - Configure Event Preference Table (EPT) in LD 117.

     For information about configuring LD 117, see *Software Input/Output Administration, NN43001-611.*.

     For information about installing and configuring an Avaya CS 1000E system, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310* .
4. Install and configure the Media Gateway Controllers.

   Note:

   Configure the Media Gateway Controllers on the Primary Call Server. The settings are included with database replication to the Secondary Call Servers.

   Figure 9: Element Manager Media Gateway configuration screen on page 56 shows the Element Manager Media Gateway configuration page.

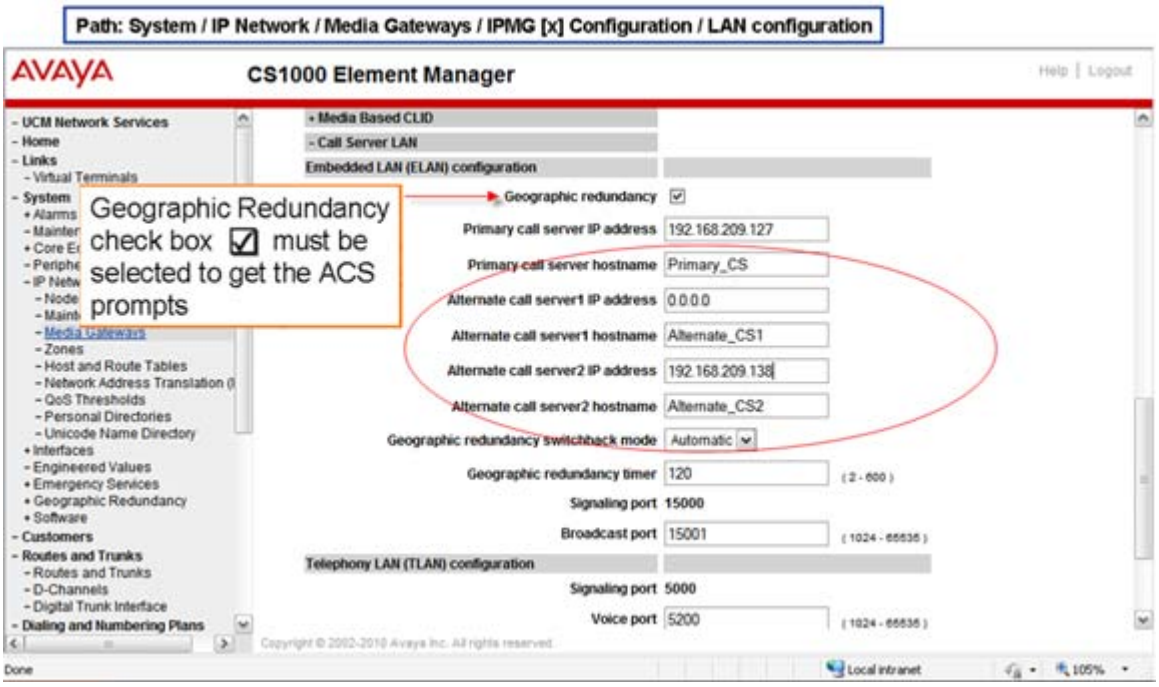

Figure 9: Element Manager Media Gateway configuration screen

5. Install and configure the Signaling Servers.

   There is an Application Node ID for secondary servers associated with each signaling server node. This application Node ID needs to be the same as the Node ID used in the Primary Call Server to define SIP signaling routes.

6. Install the Primary, Alternate and Tertiary NRSs

   The Tertiary NRS can be an actual NRS application running co-resident on the SCS signaling server or on the Survivable SIP Media Gateway media gateway signaling server.

   The Survivable SIP Media Gateway SCS and Survivable SIP Media Gateway Media gateway SIP trunk applications can also point to each other's Node IP in the Tertiary NRS configuration section for point-to-point operation without NRS registration.

   For information about configuring IP Peer Networking, see *IP Peer Networking Installation and Commissioning, NN43001-313.*

7. On the Primary system, perform a data dump (EDD) in LD 43 to create an up-to-date database files and then use the LD 43 BKR command specifying the backup rule associated with each Secondary Call Server to send the database files to the Secondary Call Server.

8. On each Secondary system, perform the following:

   - Log out from the system and wait until message SRPT4643 is displayed. This message notes the arrival of a new backup data file. Otherwise, after the manual restore and sysload, another automatic restore can occur (if ARSTR is defined) followed by an automatic sysload (if SYSLD is defined YES).

Note:

For any newly installed Secondary Call Server, the corresponding GR backup rule needs to be configured in overlay 117 to allow automatic database restore on the Secondary Call Server.

9. If required, install and configure the IP Phones with S1 pointing to the secondary node TPS and S2 pointing to the Primary system node TPS.

   For more information about installing IP Phones, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

Use the following procedure to configure a SIP Gateway on the Survivable SIP Media Gateway.

Note:

The SIP Gateway component does not require the GRSEC package.

Configuring a SIP Gateway on a Survivable SIP Media Gateway

1. Install the Call Server
   - Install the hardware. This can be a CP PM or CP MG based Co-resident Call Server and Signaling Server or a standalone CPPM Call Server.
   - Install the software.
2. Install the Media Gateway Controllers if using a CP PM card for the Call Server.

   Note:

   The MGCs are not configured on the Primary Call Server but on the local call server. The Geographic Redundancy check box should not be selected as Alternate Call Servers are not required in this configuration
3. Install and configure Signaling Servers. For CP PM/CP DC co-resident deployments additional Signaling Servers are only needed according to trunking capacity requirements.
4. Configure additional Media Cards if required, based on the number of TDM resources at the location.

   Note:

   This configuration is absent on Primary Call Server and presented on SIP GW system only.
5. Configure NRSs (if not already done while configuring the SCS portion of the Survivable SIP Media Gateway) and configure IP Peer Networking.

Configuring Database Replication on the Primary Call Server

To configure database replication on the Primary Call Server, complete the following tasks:

- Configure a Backup Rule (BKR) on page 58
- Configure a Backup Schedule (BKPS) on page 61
- Configure SECRET string for database replication on page 68
- Configure Database Replication Control (GRDRC) block on page 65
- Configure State Control (GRSC) Blocks on page 70

Configure a Backup Rule (BKR)

To perform the database-replication process, you must define a Backup Rule on the Primary system. The Backup Rule identifies the destination ELAN network interface IP address on the Secondary Call Server for the database replication. It also defines the number of versions of the database that are kept on the Secondary Call Server.

You can configure the following rule types:

- SCS = Secondary Call Server
- FMD = Fixed Media Device
- RMD = Removable Media Device
- FTP = File Transfer Protocol
- USB = Removable USB device

You can define up to 50 rules of type SCS (Secondary Call Server).You can only define one rule of type RMD (Removable Media Device) and FMD (Fixed Media Device) for each Secondary Call Server IP address. If using the FTP rule type, you must also provide a valid login and password, if required by the receiving server.

Configure the SCS backup rule as follows:

```
ld 117 new BKPR <xxx> scs <IP address> 2 <name>
```

For example: `ld 117 .new BKPR 1 scs 10.10.10.245 2 ohio`

To complete the database replication successfully, the Backup Rule must be referenced in the Database Replication Control Block. The secondary also uses the Backup Rule during the database-restore operation to identify the appropriate database to restore.

You can configure the backup rules in the CLI using LD 117 or in Element Manager.

Table 5: LD 117: Configure a Backup Rule

Command	Description
NEW/CHG BKPR xxx aaa b...b yy	Add (or change) a backup rule, where: • xxx = backup rule number ID = 1-100 (this limit increases to 512 as SCHD is configured in GRDRC block for ABKUP field). • aaa = rule type. The options for this parameter are: - RMD [<N of versions 1-10> [<name>] - FMD [<N of versions 1-10> [<name>] - FTP <IP addr><login><pwd><path>[<N of versions 1-10> [<name>] - SCS <IP addr of SCS> [<N of versions 1-10> [<name>] - USB [<N of versions 1-10> [<name>] **Note:** You must define at least one SCS rule type for each Secondary system IP address (for up to 50 Secondary Call Servers). • b...b = ELAN network interface IP address of the destination system. • yy = the number of database versions to save on the destination system = 1-(2)-10.
OUT BKPR xxx	Remove backup rule, where: • xxx = backup rule number ID = 1-100 (this limit increases to 512 as SCHD is configured in GRDRC block for ABKUP field).
PRT BKPR xxx	Print backup rule, where: • xxx = backup rule number ID = 1-100 (this limit increases to 512 as SCHD is configured in GRDRC block for ABKUP field). If no rule number is entered, all configured backup rules are printed.

For complete options regarding LD 117 commands, see*Software Input Output Reference — Administration, NN43001–611*.

Configure a Backup Rule in Element Manager

Use this procedure to configure backup rules in Element Manager.

Configuring a Backup Rule in Element Manager

1. From the Element Manager navigator menu, select **Tools > Backup and Restore > Call Server > Backup Rules**.

   The Backup Rules Web page appears.

   If no backup rules exist, you are prompted to create the first one.

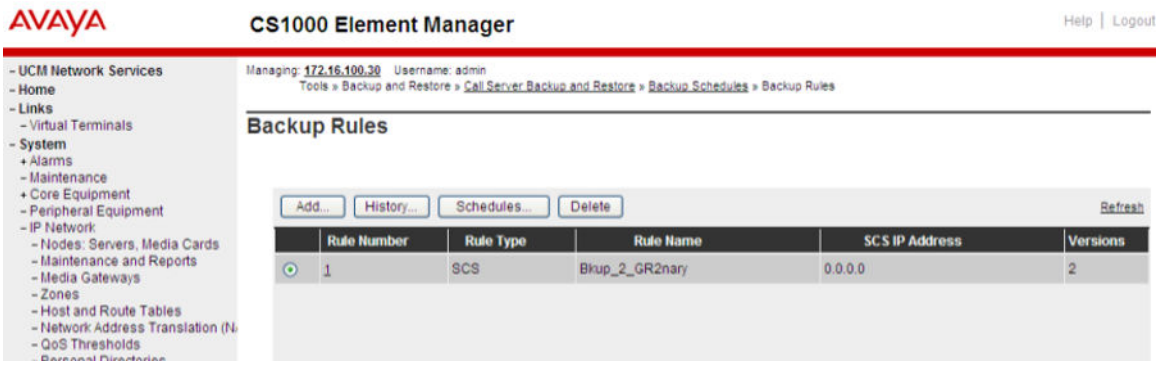

Figure 10: Element Manager Backup Rules Web page

2. To add a Backup Rule, click the **Add** button.

   The Add Backup Rule Web page appears. (To edit a Backup Rule, click the number of the rule to be edited under the Rule Number column.)

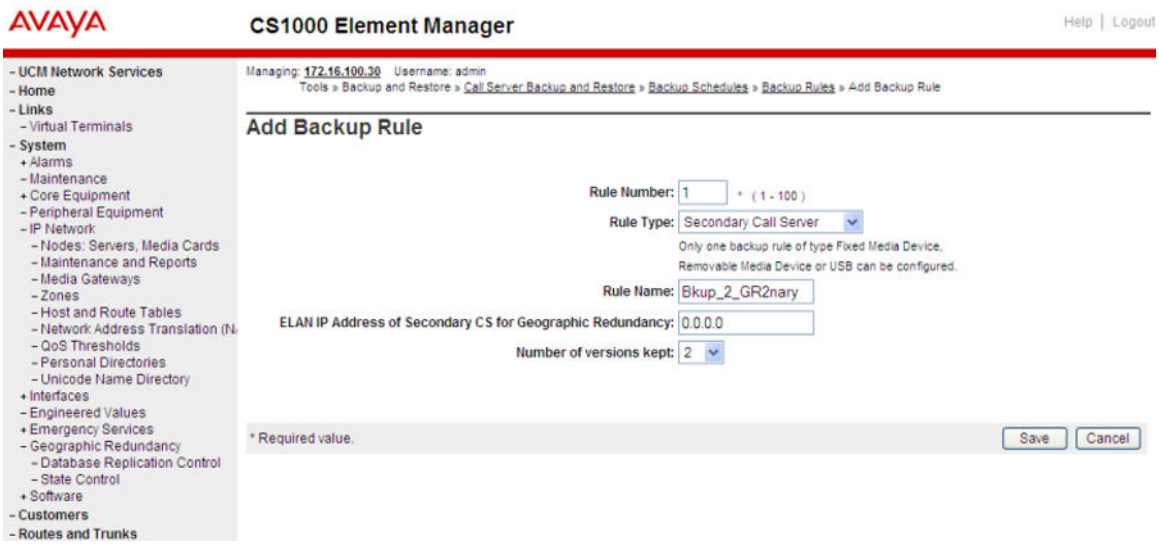

Figure 11: Element Manager Add Backup Rule Web page

3. Configure the Backup Rule as follows:

   - In the **Rule Number** box, enter the next incremental number that is available.
   - From the **Rule Type** list menu, select the appropriate rule type.
   - In the **Rule Name** box, enter a name for the rule.

- In the **IP address** box, enter the ELAN network interface IP address of the destination system for the database replication.
- In the **Number of versions kept** box, choose the number of database versions to save on the destination system.

4. Click **Save**.

Configure a Backup Schedule (BKPS)

The backup schedule allows the user to create a schedule where they can specify times for the system to initiate system backups. Backup schedules can be created using the CLI in LD 117 or in Element Manager.

Note:

Skip this step if there is more than one SCS. Use the GRNS command instead. The GRNS command must be used after the GRSC and GRDRC commands. If it is not, the following error is displayed: `-> new grns SCH2198 GRNS commands are available only for GR N-way`.

Table 6: LD 117: Configure a Backup Schedule

Command	Description
NEW/CHG BKPS xxx yyy [<FREQ><DAY><HOUR><MINUTE>]	
	Add (or change) a new backup schedule, where: • xxx = backup schedule number ID = 1-10 (upper limit increases to 512 if SCHD is configured in GRDRC block for ABKUP field). • yyy = number of the backup rule for the scheduled backup operation. • FREQ = defines how often the backup takes place. The options for this parameter are: - M = monthly - W = weekly - D = daily - Triggered by EDD = Triggered by Equipment Data Dump (this option is not available when Automatic Backup field is configured as ABKP = SCHD). • DAY = This parameter is dependant upon the entry for FREQ. - If M was selected for FREQ, this value is (1)-31. - If W was selected for FREQ, this value is one of SU / MO / TU / WE / TH / FR / SA.

Command	Description
	- If D was selected for FREQ, this value is represented by the next parameter, HOUR. • HOUR = 0-(3)-23. • MINUTE = (0)-59.
OUT BKPS xxx	Remove backup schedule, where: • xxx = backup schedule number ID = 1-10 (upper limit increases to 512 if SCHD is configured in GRDRC block for ABKUP field).
PRT BKPS xxx	Print backup schedule, where: • xxx = backup schedule number ID = 1-10 (upper limit increases to 512 if SCHD is configured in GRDRC block for ABKUP field).

Note:

You must invoke MIDN from LD 135 to test the SCHD block.

Configure a Backup Schedule in Element Manager

From the Backup Schedules Web page, you can do the following:

- Add a new Backup Schedule
- Edit a Backup Schedule
- Configure Automatic Schedules
- Delete a Backup Schedule

To add a backup schedule in Element Manager, do the following:

Adding a Backup Schedule in Element Manager

1. From the Element Manager navigator menu, select **Tools > Backup and Restore > Call Server > Backup Schedules**.

   The Backup Schedules Web page appears.

   If no backup schedules exist, you are prompted to create the first one.

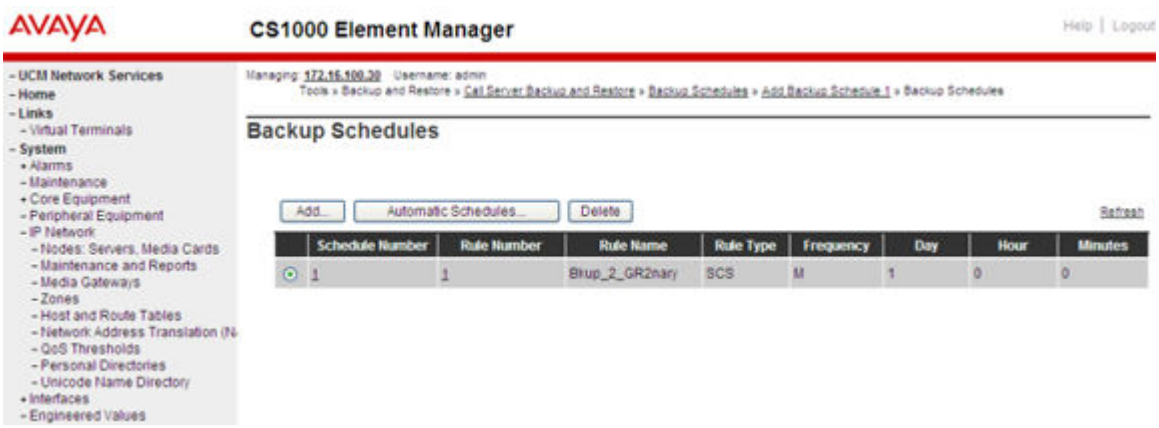

Figure 12: Element Manager Backup Schedules Web page

2. Click **Add** .

   The **Add Backup Schedule** Web page appears.

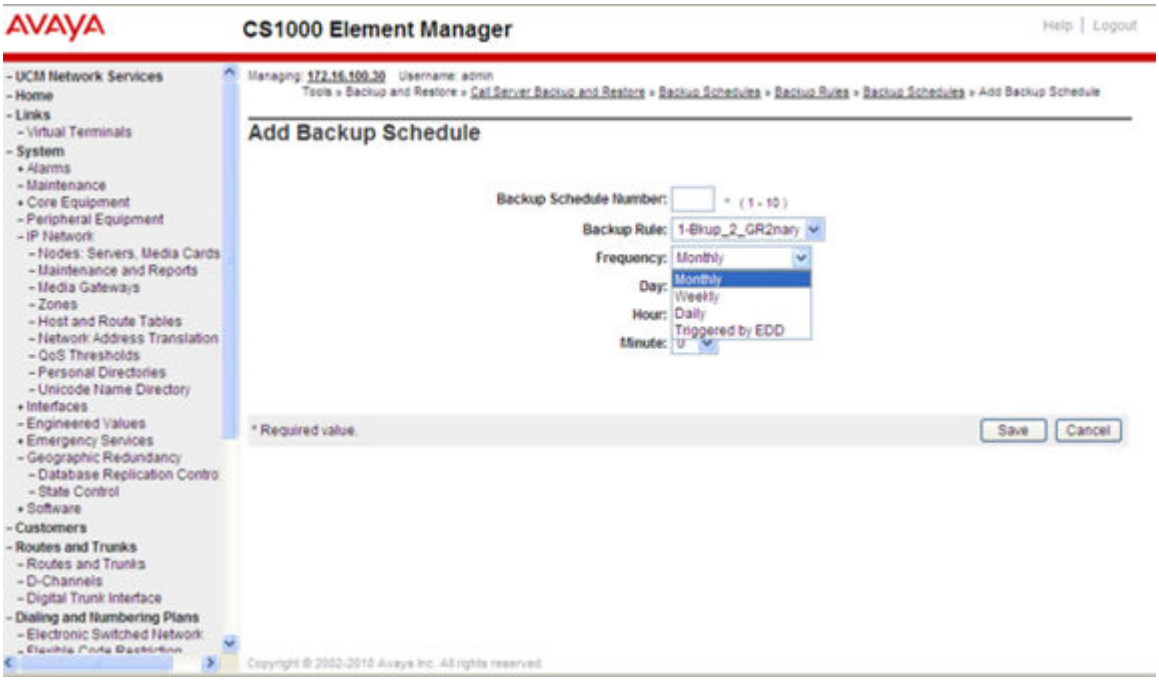

Figure 13: Element Manager Add Backup Schedule Web page

3. Configure the backup schedule as follows:

   - In the **Backup Schedule Number** box, enter the schedule number. If Automatic Replication Backup is defined as SCHD during GRDRC configuration, the range is 1 to 512. Otherwise, the default range is 1 to 10.
   - Choose the appropriate value from the **Backup Rule** drop-down list.

- Choose the appropriate values from the **Frequency**, **Day**, **Hour**, **Minute**, and **Triggered by EDD** lists as required.

4. Click **Save**.

To edit a backup schedule, click the number of the rule to edit under the Schedule Number column.

To configure Automatic Schedules, do the following:

Note:

Automatic Schedules are only applicable for Backup Rules of type SCS.

Configuring Automatic Schedules in Element Manager

1. From the Backup Schedules Web page, click **Automatic Schedules**.

   The Update Backup Schedules Web page appears.

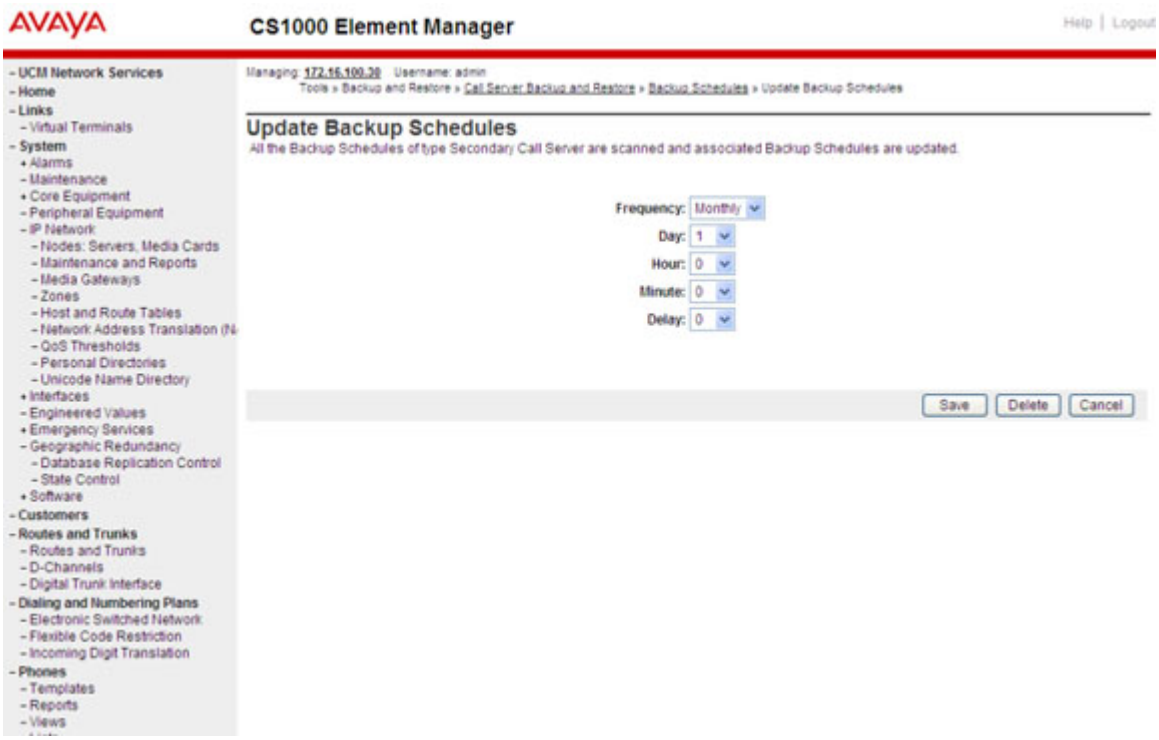

Figure 14: Element Manager Update Backup Schedules Web page

2. From the lists, choose the desired **Frequency**, **Day**, **Hour**, **Minute**, and **Delay** settings.
3. Click **Save**.

   All SCS backup rules are updated with the new settings.

GRNS command

The GRNS command simplifies the creation of backup schedules by scanning the Secondary Call Server backup rules and configuring the BKPS for each of them, adjusting the start times according to the specified delay value. Use the GRNS command as an alternative to the configuration of automatic backup schedules in Element Manager.

You can issue the GRNS command from LD 117.

Table 7: LD 117: GRNS command

Command	Description
NEW/CHG GRNS <FREQ><DAY><HOUR><MINUTE>[<DELAY>]	
	Add (or change) a Survivable Media Gateway Backup, where: • FREQ = how often the backup takes place, where: - M = monthly - W = weekly - D = daily • DAY = this parameter is dependant upon the entry for FREQ, where: - if FREQ = M, the value is (1)-31. - if FREQ = W, the value is one of (SU), MO, TU, WE, TH, FR, or SA. - if FREQ = D, the next parameter is HOUR. • HOUR = 0-(3)-23 • MINUTE = (0)-59 • DELAY = 0-(3)-60 This value represents the interval in minutes between two consecutively scheduled backups. The system scans for backup rules of SCS type and creates (or modifies) a BKPS for each of them, adjusting the start times according to the specified delay value.
OUT GRNS	This command removes all backup schedules that reference backup rules of type SCS.

Configure Database Replication Control (GRDRC) block

The database replication process requires that you configure a Geographic Redundancy Database Replication Control (GRDRC) block on the Primary system. The GRDRC block

defines how the database replication is initiated on the Primary system and if the restore and sysload operations are performed automatically or manually on the Secondary system.

In a Survivable Media Gateway system, only one GRDRC block is configured independent of the total number of Secondary Call Servers assigned to the system. You must configure the ABKUP parameter as SCHD, with the backup schedule for each Secondary Call Server configured according to its associated backup rule. You can apply any valid Secondary Call Server backup rule from the list but the rule must have a backup schedule associated with it. Otherwise, the system will not function as a Survivable Media Gateway configuration.

You can configure the GRDRC block using LD 117 or Element Manager.

Important:

You must configure ABKUP = SCHD. Configuring ABKUP with any other value causes the Survivable Media Gateway configuration to work as a 1+1 configuration.

Table 8: LD 117: Configure GRDRC block

Command	Description
NEW/CHG GRDRC xxx aaa yyy bbb ccc	
	Add (or change) a GRDRC block, where: • xxx = Backup Rule number. • aaa = how the automatic database replication to the destination system occurs. (Geographic Redundancy requires that this parameter be configured as SCHD): - SCHD—according to defined backup schedule - IMM—immediately after any data dump operation - MIDN—after midnight data dump only - NO—not allowed • yyy = Restore Rule = Backup Rule number used for the restore operation. • bbb = (YES)/NO. Defines whether or not the automatic restore operation (ARSTR) is allowed. • ccc = (YES)/NO. Defines whether or not the automatic sysload after successful automatic restore (ASYSLD) is allowed. ccc = YES is only allowed if bbb = YES.
OUT GRDRC	Remove current GRDRC Block.
PRT GRDRC	Print GRDRC Block.

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Configure GRDRC block in Element Manager

To configure the GRDRC block in Element Manager, perform the following procedure.

Configuring GRDRC block in Element Manager

1. From the Element Manager navigator menu, select **System > Geographic Redundancy > Database Replication Control**.

   The Database Replication Control Web page appears.

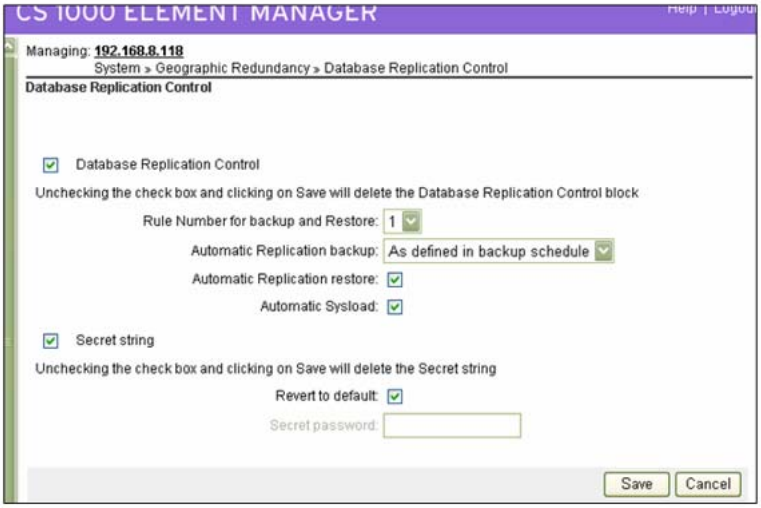

   Figure 15: Element Manager Database Replication Control Web page

   If no backup rule of type SCS is configured, a message displays asking if you want to configure one. Clicking **Ok** opens the Backup Rules page. GRDRC options are not available for configuration unless a backup rule of type SCS exists.

   If no GRDRC block has been configured, a message appears confirming that you want to create one.

2. Click **Ok**.
3. Select the **GRDRC** check box.

   The GRDRC configuration options are available (not dimmed).

4. Select the desired **Rule Number for Backup and Restore** .

   You can apply any valid Secondary Call Server backup rule from the list, providing that each backup rule also has a backup schedule associated with it.

5. Choose the **Automatic Replication Backup** value from the list.
6. To enable the automatic restore operation, select the **Automatic Replication Restore** check box.

7. To allow the automatic sysload operation, select the **Automatic Sysload** check box.
8. Click **Save**.
9. To delete a GRDRC block, clear the GRDRC check box and click **Save**.

   The GRDRC block is deleted.

Configure SECRET string for database replication

Note:

The SECRET string is only used for database encryption and decryption during upgrades of the Primary Call Server to Communication Server 1000 Release 6.0 from Communication Server 1000 Release 5.x.

The SECRET string is a pre-shared security password enabling the Secondary Call Servers to decrypt the database sent from the Primary Call Server. Each Call Server in the system must have the same SECRET string password for database replication to be successful. The SECRET string does have a default on both systems which allows it to work. It is recommended to change this string on both systems for secure encryption of the transferred data.

The SECRET string is configured using LD 117.

Note:

The SECRET string must be configured on the Primary Call Server and all Secondary Call Servers individually. It is not replicated to the Secondary Call Servers during database replication.

Table 9: LD 117: Configure SECRET string for database replication

Command	Description
SECRET SET GR <applName> <secret>	Define the SECRET string. The following limitations apply: • Maximum of 30 characters. • Cannot use the "!" character.
SECRET DEFAULT GR	Restore the SECRET string to the default value.
SECRET STAT GR	Print the SECRET string.

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Configure SECRET string in Element Manager

Note:

The SECRET string is only used for database encryption and decryption during upgrades of the Primary Call Server to Communication Server 1000 Release 6.0 from Communication Server 1000 Release 5.x.

The SECRET string is a pre-shared security password enabling the Secondary Call Servers to decrypt the database sent from the Primary Call Server. Each Call Server in the system must have the same SECRET string password for database replication to be successful.

To configure the SECRET string in Element Manager, perform the following procedure.

Configuring SECRET string in Element Manager

1. From the Element Manager navigator menu, select **System > Geographic Redundancy > Database Replication Control**.

   The Database Replication Control Web page appears.

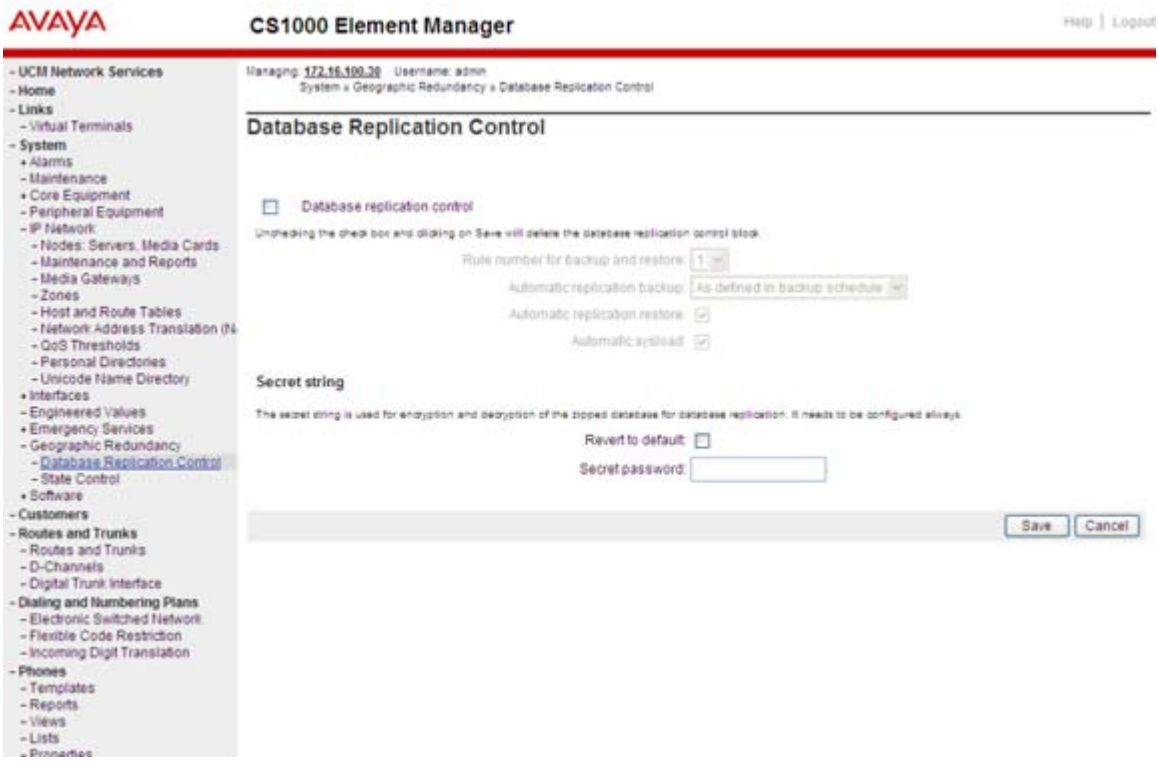

Figure 16: Element Manager Database Replication Control Web page

2. Configure the SECRET string in one of the following ways:

   a. To configure the SECRET string with the default password, select the **Revert to default** check box. Click **Save**.

      The SECRET string is saved with the default value.

Or

b. To configure a new SECRET string value, clear the **Revert to default** check box. Enter a new value in the **Secret Password** box. The new password must conform to the following limitations:

   • Maximum of 30 characters.

   • The "!" character cannot be used.

   Click **Save**.

   The SECRET string is configured with the new value.

3. To remove the SECRET string value, clear the **Secret String** check box and click **Save**.

   The SECRET string value is removed.

Configure State Control (GRSC) Blocks

The Geographic Redundancy State Control (GRSC) block specifies the Secondary system state control parameters that allow the system to transition from INACTIVE to ACTIVE states. For more information, see Secondary system operating states on page 36.

Each Secondary Call Server must have its own GRSC block configured. The system supports up to 512 configured GRSC blocks.

Table 10: LD 117: Configure GRSC blocks

Command	Description
NEW/CHG GRSC <associated scs>[<GRTHR1><GRTHR2><STFTO><FCTO><SDAM>]	
	Add (or change) a new GRSC block, where: • associated scs—Secondary Call Server to be associated with the GRSC block, where: - Backup Rule Number—number of the SCS backup rule configured. - NBKP—defines a generic GRSC block for Secondary Call Servers that do not have an associated backup rule configured. - ALL—automatically configures GRSC blocks for all Secondary Call Servers that have associated backup rule defined. • GRTHR1—the number (N) of IP Phones that must register on the Secondary system for the system to escalate to the ACTIVATING state. If no value is entered, xxx = 1. The maximum value of xxx is: 10% x (Basic IP User license + IP User license). • GRTHR2—the defined threshold for the number of Media Gateways registered to the Secondary Call Server. If the

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Command	Description
OUT GRSC <associated scs>	Secondary system is INACTIVE and the number of Media Gateways registered exceeds GRTHR2, the system escalates to the ACTIVATING state. • STFTO—Short Term Failure Timer, in minutes = (5) - 600. • FCTO—Failure Clearance Timer, in minutes = (5) - 180. • SDAM—Secondary system Deactivation Mode = (AUTO)/ MAN. Remove GRSC Block, where: • <associated scs>—Secondary Call Server to be associated with the GRSC block, where: - Backup Rule Number—number of the SCS backup rule configured. - NBKP—defines a generic GRSC block for Secondary Call Servers that do not have an associated backup rule configured. - ALL—automatically configures GRSC blocks for all Secondary Call Servers that have associated backup rule defined.
PRT GRSC [(ALL)/ <associated scs backup rule>/NBKP]	Print GRSC Block, where: • ALL—all Secondary Call Servers that have backup rules defined. • associated scs backup rule—number of the SCS backup rule configured. • NBKP—a generic GRSC block for Secondary Call Servers that do not have an associated backup rule configured.

Configure GRSC block in Element Manager

To configure the GRSC block in Element Manager, do the following:

Configuring GRSC block in Element Manager

1. From the Element Manager navigator menu, select **System > Geographic Redundancy > State Control**.

   The State Control Web page appears.

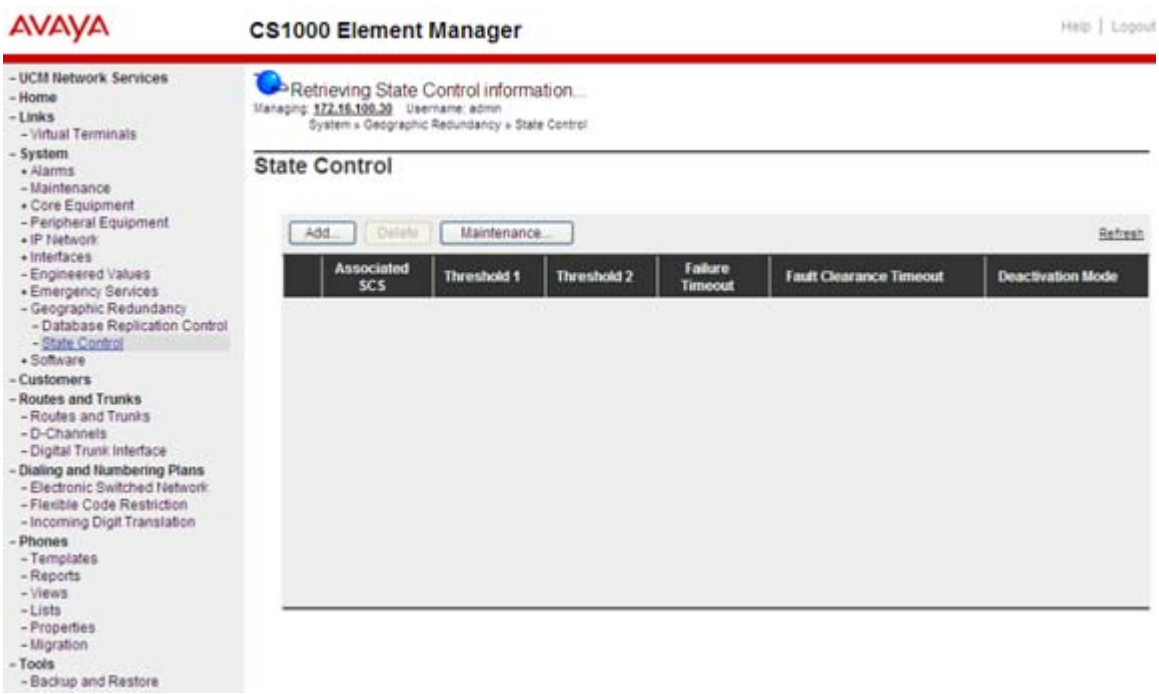

Figure 17: Element Manager State Control Web page

2. To add a new GRSC block, click the **Add** button.

   Note:

   A backup rule of type SCS must exist before new GRSC blocks can be added.

   To edit a GRSC block, click the link under the Associated SCS column.

   The Add State Control Web page appears.

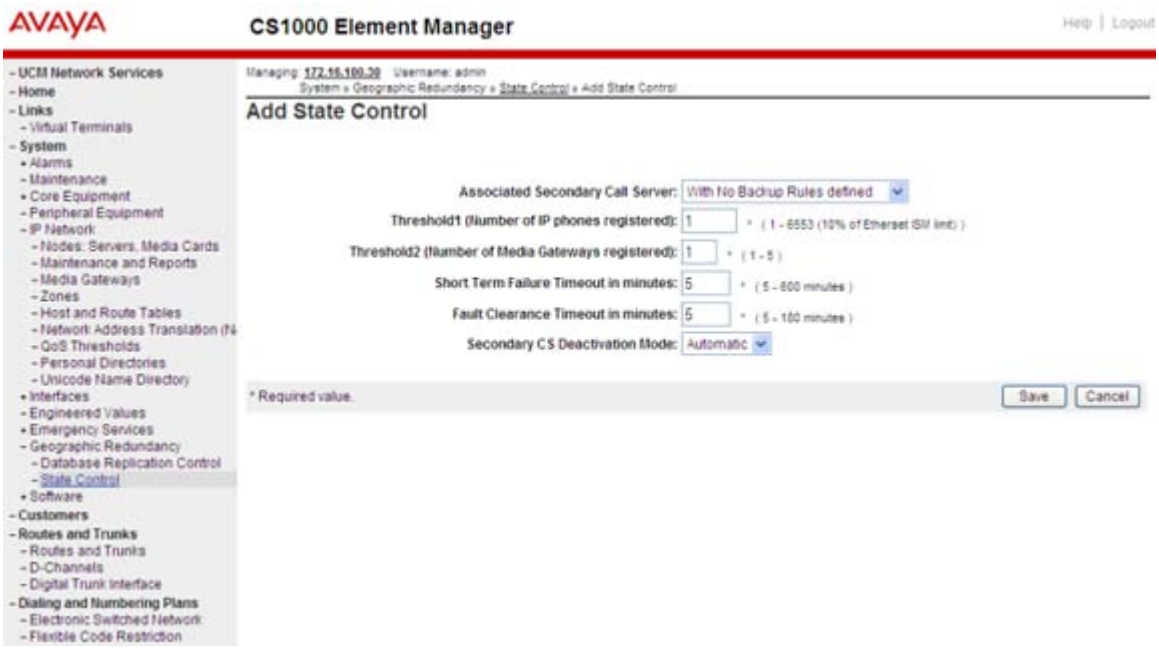

Figure 18: Element Manager Add State Control Web page

3. Choose the appropriate item from the **Associated Secondary Call Servers** list.
4. Enter the desired IP Phone threshold number in the **Threshold1** box.
5. Enter the desired Media Gateway threshold number in the **Threshold2** box.
6. Enter the number of minutes in the **Short Term Failure Timeout** box.
7. Enter the number of minutes in the **Fault Clearance Timeout** box.
8. Choose the deactivation mode from the **Secondary CS Deactivation Mode** list.
9. Click **Save**.

Configure the Media Gateway Controller (MGC)

Package 404 or 405 must be enabled to configure the Media Gateway Controller in Element Manager.

Note:

Configure the Media Gateway Controllers on the Primary Call Server. The settings are included with database replication to the Secondary Call Servers.

To configure the Media Gateway Controller in Element Manager, do the following:

Configuring the Media Gateway Controller in Element Manager

1. From the Element Manager navigator menu, select **System > IP Network > Media Gateways**.

The Media Gateways Web page appears.

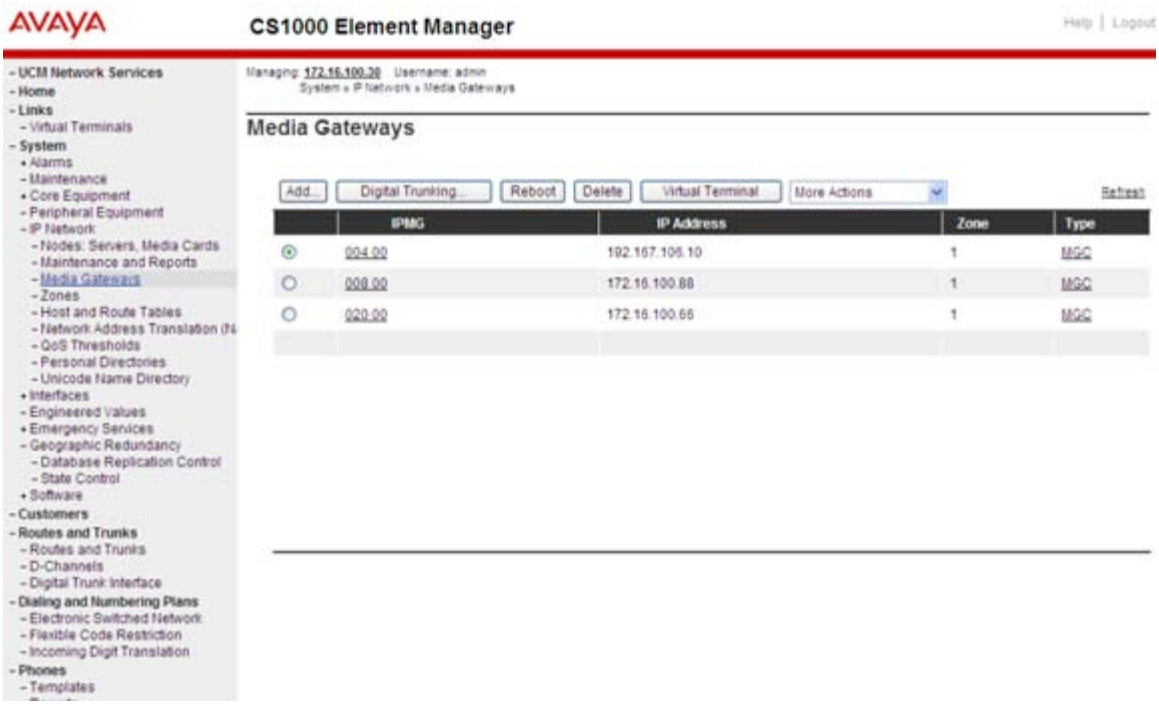

Figure 19: Element Manager Media Gateways Web page

2. From the list of IPMGs, click the MGC link (under the Type column) for the device to be configured.

   The IPMG 40 0 Property Configuration Web page appears, as shown in Figure 20: Element Manager IPMG 40 Property Configuration Web page on page 75.

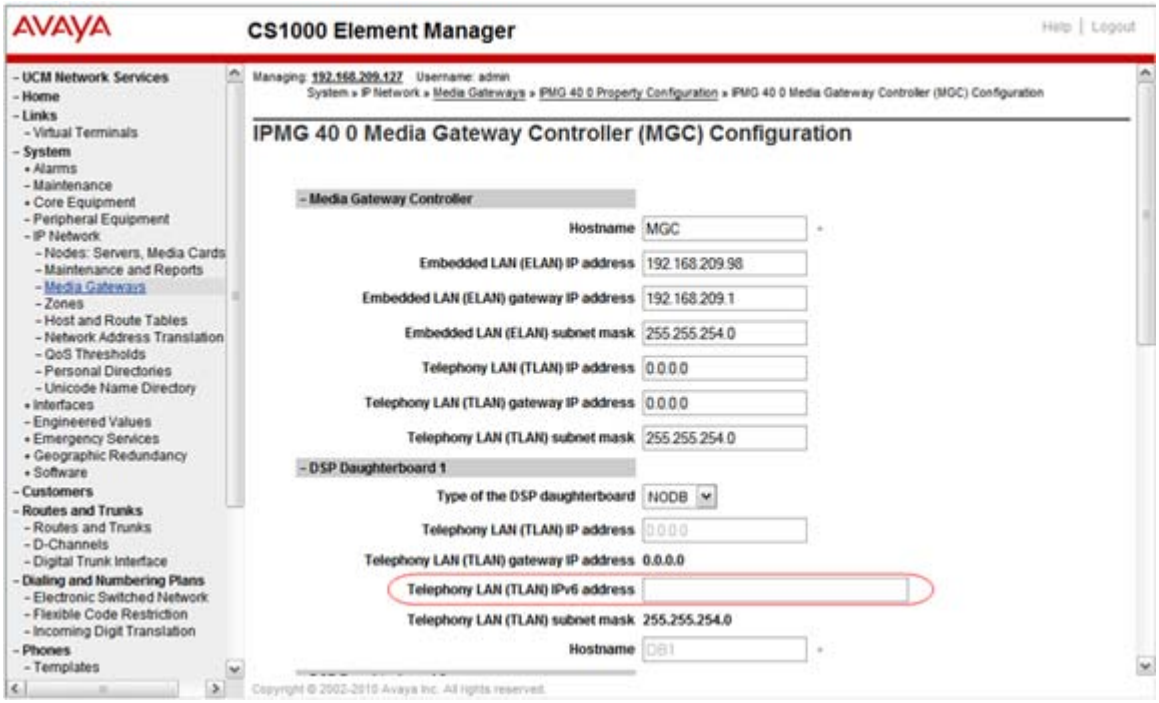

Figure 20: Element Manager IPMG 40 Property Configuration Web page

3. Click [+] to expand the LAN configuration section.

   If Package 404 is enabled, the Primary Call Server IP is identical to the Call Server that Element Manager is managing and appears dimmed (read-only).

4. Enter IP values for the Primary Call Server, Alternate Call Server 1, and Alternate Call Server 2 in the appropriate boxes.

   Note:

   With a Survivable SIP Media Gateway component the MGC should be registered with the SIP GW component, not the Primary Call Server.

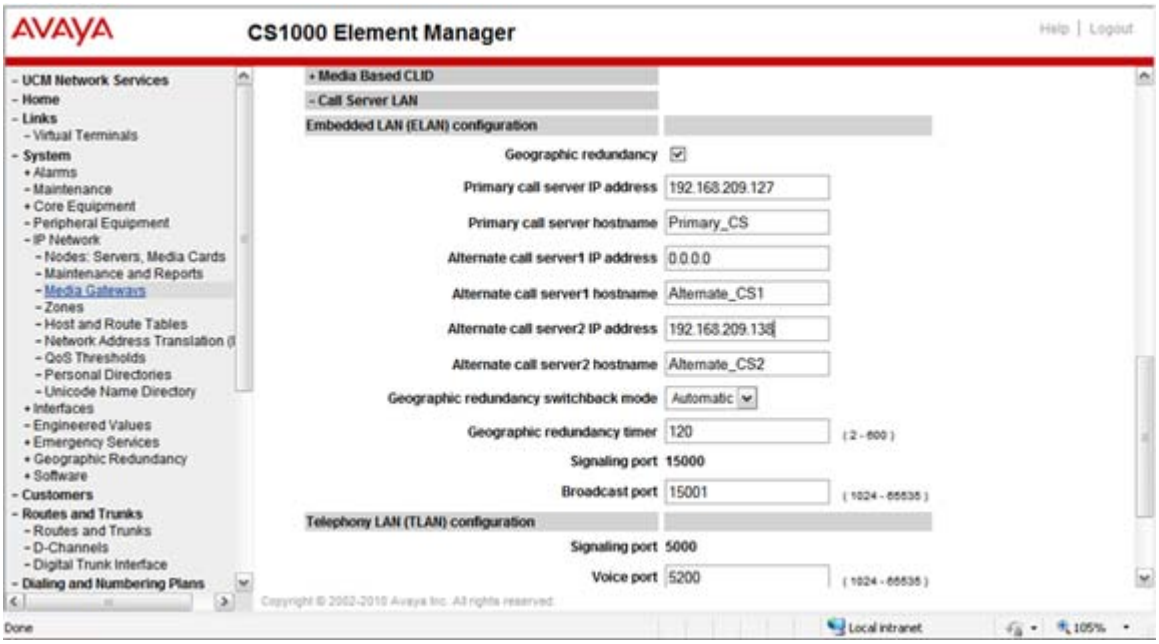

Figure 21: Element Manager IPMG 40 0 Property Configuration Web page, continued

5. Fill in any other mandatory boxes (mandatory boxes are marked with an asterisk [*]).
6. Click **Submit**.

For more information about the installation and configuration of the Media Gateway Controller, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

For a complete list of available Media Gateway Controller commands, see *Avaya Software Input/Output Administration, NN43001-611*.

Configure IP telephony nodes

Package 404 or 405 must be enabled to configure IP telephony nodes in Element Manager. To configure IP telephony nodes in Element Manager, do the following:

Configuring IP telephony nodes in Element Manager

1. From the Element Manager navigator menu, select **System > IP Network > Nodes: Servers, Media Cards**.

   The Node Configuration Web page appears.

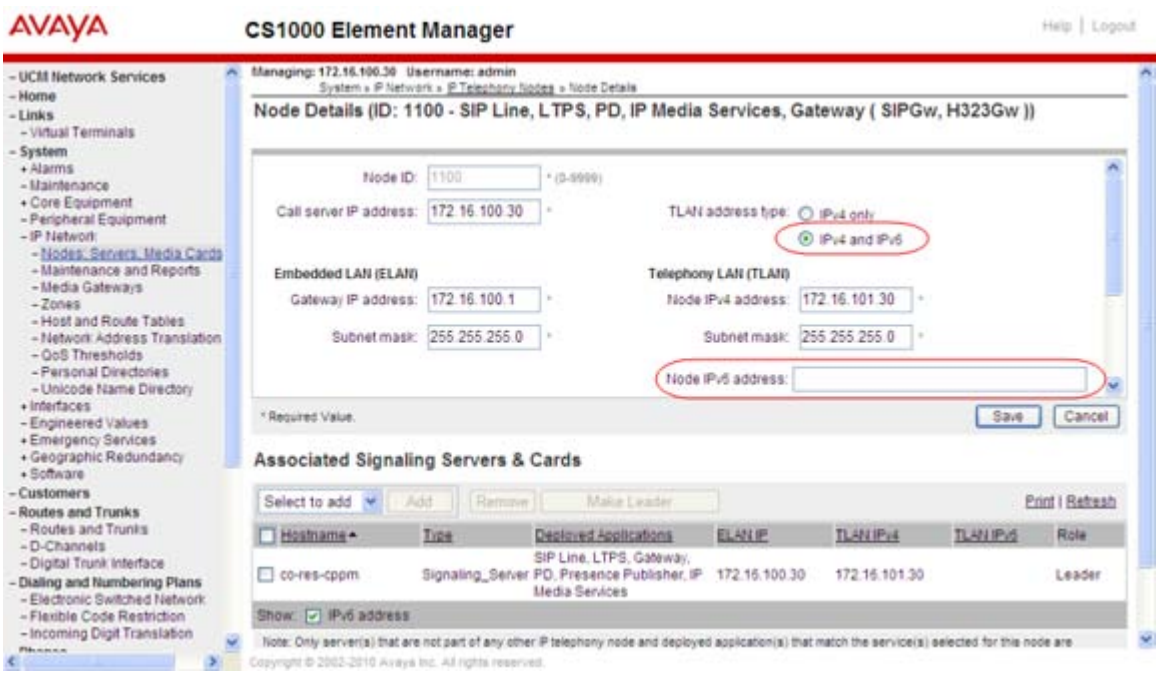

Figure 22: Element Manager Node Configuration Web page

2. Select **Edit** for an existing node.

   The Edit Web page appears.

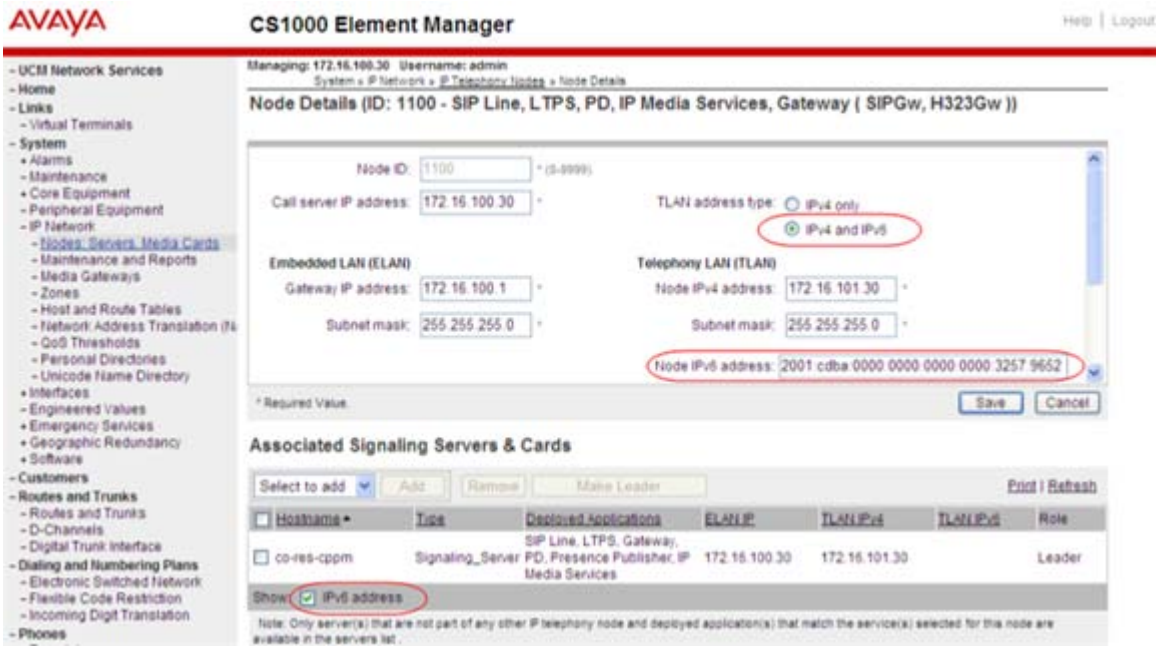

Figure 23: Element Manager Edit Node Configuration Web page

3. From the drop-down list under the **Application Signaling Servers & Cards** section, select the Card to display the Card properties.

4. Select the **Card processor type** from the list. (See Figure 8: Element Manager Media Card configuration screen on page 51.)

   If Pentium Card is selected, the IP fields for Primary Call Server, Alternate Call Server 1, and Alternate Call Server 2 are not displayed.

   If Voice Media Gateway or MC 32S card are selected, the IP fields for Primary Call Server, Alternate Call Server 1, and Alternate Call Server 2 are displayed.

   If Package 404 is enabled, the Primary Call Server IP is identical to the Call Server that Element Manager is managing and appears dimmed (read-only).
5. Enter the IP values for the Primary Call Server, Alternate Call Server 1, and Alternate Call Server 2 in the appropriate boxes.
6. Fill in any other mandatory boxes (mandatory boxes are marked with an asterisk [*]).
7. Click **Save**.

Configure Primary (S1) and Secondary (S2) Connect Servers

To install and configure IP Phones, refer to *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*. Ensure the S1 of each IP Phone points to the local Secondary system node TPS, and the S2 points to the Primary system node TPS. Refer to Figure 33: Simple redundancy on page 105 and Figure 34: Multiple site redundancy on page 106 for examples of how S1 and S2 are defined.

For information about configuring IP Phones using DHCP, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Adding a new Secondary Call Server to the system

A total of 511 Secondary Call Servers can be added the system, if they are all configured as Survivable SIP Media Gateways. A total of 50 Secondary Call Servers can be added the system, if they are all configured as Survivable Media Gateways. If all the Secondary Call Servers in the Geographic Redundant network are Survivable Media Gateways, then the limit of 50 Secondary Call Servers holds. In a mixed Survivable SIP Media Gateway and Survivable Media Gateway deployment, there can be a total of 511 Secondary Call Servers, of which only 50 can be Survivable Media Gateways. To add a Secondary Call Server to the system, do the following:

Adding a new Secondary Call Server to the system

1. On the new Secondary Call Server:
   - Install the hardware (if required).

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- Install CS 1000 Release 6.0 (same version as Primary or later).
- Configure the Call Server with system specific data, such as IP address, Signaling Server settings, and so forth.

For more information about installing and configuring an Avaya CS 1000E system, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

2. On the Primary Call Server, configure the following for each new Secondary Call Server:

   - Backup Rule (see Configure a Backup Rule (BKR) on page 58).
   - Backup Schedule (see Configure a Backup Schedule (BKPS) on page 61).
   - State Control Block (see Configure State Control (GRSC) Blocks on page 70).

   For each Secondary Call Server, use the BKR command in LD 43 (specifying the associated backup rule) or use the Backup According to Rule in Element Manager to replicate the database. For more information, see Manual database replication and restore on page 85.

3. On the new Secondary Call Server:

   - After replicating the backup set from the Primary Call Server, define Backup Rule 1. For more information, see Configure a Backup Rule (BKR) on page 58.
   - Define the GRDRC data block as follows:

     - Rule for BKUP = 1
     - Rule for Restore = 1
     - ARSTR = NO
     - ASYSLD = NO
     - ABKUP = SCHD

     Note:

     If you do not define the GRDRC block on the Secondary Call Server before performing the first manual database replication, the IP addresses for the IPMGs are not restored.

     For more information about configuring the GRDRC block, see Configure Database Replication Control (GRDRC) block on page 65.

   - In LD 43, use the RSR command to manually restore the database from the Primary Call Server backup set. (Filtering of Primary system specific data must be used.)
   - Sysload the Call Server.

Adding a new Secondary Call Server in the Survivable SIP Media Gateway configuration

1. On the Secondary Call Server component of the Survivable SIP Media Gateway, install a Secondary Call Server
   - Install hardware. This can be a CP PM or CP DC based Co-resident Call Server and Signaling Server or a standalone CP PM Call Server.
   - Install the software.
   - Configure the Call Server with system specific data, such as IP address and Signaling Server settings.
   - Configure the corresponding GR backup rule to this new Secondary server.
2. On the Primary Call Server, configure the following for each new Survivable SIP Media Gateway
   - Backup Rule
   - Backup Schedule
   - State Control Block

     For each Secondary Call Server , use the LD 43 BKPR command.
3. On the SCS component of the SSMG • • In LD43 use the RSR command to manually restore the database from the Primary Call Server backup set • Sysload the call server
   - After replicating the backup data from the Primary Call Server, define a backup rule 1.
   - In LD 43 use the RSR command to manually restore the database from the Primary Call Server backup set.
   - Sysload the Call Server.
4. On the SIP Gateway component of the SSMG, install a CS1000 Call Server
   - Install hardware (CPPM platform to run the CS/SS applications or the CPMG platform)
   - Install the Media Gateway Controller card for media services if the CS is on a CPPM. If the CS platform is a CPMG, then there is not need for an MGC card. Install additional Media Cards (SMC or ITG-SA) if more DSPs are required.
5. Configure NRSs (if not already done while configuring the SCS portion of the SSMG) and configure IP Peer Networking)

Recovering a modified Secondary Call Server database

During normal operation, all customer database changes are performed on the Primary Call Server only and the database is replicated to the Secondary Call Servers. If the Primary Call

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Server is offline for an extended period of time (i.e. many days), it may be necessary to continue making database changes on the Secondary Call Server(s).

Although not a normal process, a Secondary Call Server database can be replicated back to the Primary Call Server to become the new primary image. Only an image from one Secondary Call Server can be recovered, so this process is only useful if there is only one or a few Secondary Call Servers.

When the Secondary system is in the ACTIVE state following Primary system failure, administrative changes made on the Secondary system can be preserved.

Before making any changes, define ARSTR = NO in the GRDRC block (see Configure Database Replication Control (GRDRC) block on page 65). This prevents the automatic restore process from accidentally overwriting the latest changes made on the Secondary system.

The Secondary system administrative changes can be saved by performing a data dump.

Clear ACTIVE state

If the Secondary system is in the ACTIVE state and the Secondary system Deactivation Mode is defined as Manual (SDAM = MAN) in the GRSC block, clear the Secondary system ACTIVE state in LD 135 by entering the following on the Secondary system:

Table 11: LD 135: Clear Geographic Redundancy

Command	Description
CLR GR	Clear Secondary system ACTIVE state. This command triggers repetitive attempts to redirect all the IP Phones to the Primary system for a maximum period defined by FCT. If the Primary system is operational and N1 falls below GRTHR1 and N2 falls below GRTHR2, the system transitions to the DEACTIVATING state. It also sends a switch to Primary request to all Media Gateway Controllers registered on the Secondary Call Server.

Clear ACTIVE state in Element Manager

To clear the Secondary system ACTIVE state in Element Manager, do the following.

Clearing Secondary system ACTIVE state in Element Manager

1. In the Element Manager navigator tree, select **System > Maintenance > Select by Overlay > LD 135—Core Common Equipment Diagnostics > Geographic Redundancy Diagnostics**.

   The Geographic Redundancy Diagnostics Web page appears.

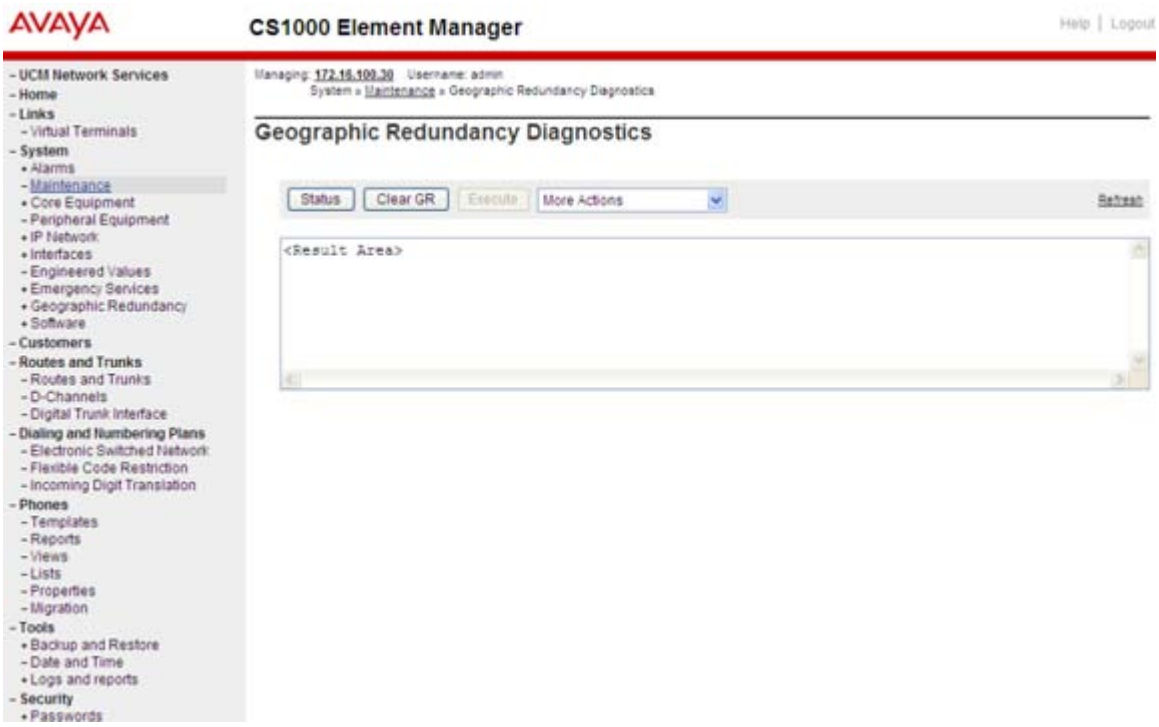

Figure 24: Element Manager Geographic Redundancy Diagnostics Web page

2. Click the **Clear GR** button.

   The ACTIVE state for the Secondary Call Server is cleared.

Primary system recovery

When the Primary system experiences a long-term failure, any administrative changes made on the active Secondary system during this period are saved on the Secondary system only. These changes can be preserved and replicated to the Primary system after it returns to normal operations.

Recovering the Primary system database on page 82 describes how to perform a system recovery following Primary system failure.

Recovering the Primary system database

1. On the Secondary system:

   a. Define a new Backup Rule of type FTP in LD 117 to backup the Secondary Call Server database to an external FTP server. For more information, see Configure a Backup Rule (BKR) on page 58.

   b. Perform a data dump in LD 43. (As the Secondary system is equipped with the GRSEC package, this does not automatically initiate a backup to the Primary system.)

c. In LD 43, enter:

BKR *<BKUP rule>*

- *<BKUP rule>* is the Backup Rule defined in 1a.

This manually initiates the database replication to the external FTP server. For more information, see Manual database replication and restore on page 85.

2. On the Primary system:

a. Define a new Backup Rule of type FTP in LD 117 for the external FTP server. For more information, see Configure a Backup Rule (BKR) on page 58.

b. In LD 43, enter the following:

RSR *<RULE for Restore>*

- *<Rule for Restore>* is the Backup Rule defined in 2a. System-specific data for the Primary system is filtered.

This manually restores the database that was transferred to the external FTP server.

c. Perform a sysload to accept the newly restored data.

Testing the recovered Primary system on page 83 describes how to perform a system test following Primary system recovery.

Testing the recovered Primary system

1. On the Primary system:

a. In LD 117, enter the following:

CHG GRDRC *<BKUP rule>* **IMM** *<Rule for Restore>* **YES YES**

where

<BKUP rule> and *<Rule for Restore>* are both the original Backup Rule for replication to the Secondary system (not the rule defined in Recovering the Primary system database on page 82.

b. Perform a data dump in LD 43.

The data dump initiates the automatic database replication to the Secondary system.

2. On the Secondary system:

a. In LD 117, define ARSTR = YES and ASYSLD = YES in the GRDRC data.

b. Log out of the system and wait for the automatic restore and automatic sysload to take place, then accept the backup database.

Upgrades

If a system upgrade or patch is required, both the Primary and Secondary systems must be upgraded to have the matching software issue. The upgrades must be performed separately at the Primary and Secondary sites. Upgrade or patch the Secondary site first, followed by the Primary site.

Upgrade or install the software on the Secondary site Call Server and Signaling Server, apply any applicable patches, and then upgrade or install the software and patches on the Primary system. This allows the database replication process to function normally during the upgrades. The Secondary system performs an automatic conversion of older database versions that arrive while the Primary system is temporarily running the older software.

Note:

If the Primary system attempts to transfer a database version that is higher than the Secondary system software, the automatic restore is blocked and a warning message is printed on the TTY.

The Primary and Secondary systems conform to existing methodologies to deploy maintenance and diagnostic patches.

Maintenance

The Primary and Secondary systems conform to the methods and procedures used for local and remote access as implemented in Avaya CS 1000 Release 6.0 software. All systems also conform to the existing methodologies used to deploy maintenance and diagnostic patches as implemented in Avaya CS 1000 Release 6.0 software.

Survivable Media Gateway preserves all existing overlay support tools available in Avaya CS 1000 Release 6.0 software for Avaya CS 1000E systems, except where otherwise noted in this document.

Element Manager

Element Manager (EM) can be used to administer the Primary and Secondary Call Servers in a Survivable Media Gateway configuration.

Avaya recommends that you do configuration changes on the Primary Call Server using EM. These changes are automatically copied to the Secondary Call Servers by the Geographic Redundancy replication process. Any changes made directly on the Secondary Call Servers will typically be overwritten on the next replication.

Database configuration

During normal operation, the Primary system is the master of the customer database. Perform all database modifications on the Primary system only.

Manual database replication and restore

The database replication and restore procedures are initiated using LD 43.

Table 12: LD 43: Manual database replication

Command	Description
BKR xx	Invoke database-replication operation, where: • xx = Backup Rule number. This command is typically entered on the Primary system for replication to the Secondary system.
RSR xx yy	Restore the database, where: • xx = Restore Rule number on the local system. This value must be the same as the Restore Rule defined in the local system GRDRC block; otherwise, all system-specific definitions are lost. • yy = database version number. If no version number is entered, the most recent backup (1) is used. The latest database version is assigned the highest priority. For example: yy = 1 restores the latest backup database; yy = 2 restores the second latest database version. This command is typically entered on the Secondary system to restore a database received from the Primary system. When the RSR command is issued, the following confirmation prompt appears: • **`Perform IP Config filtering (Y / N)?`** If the user enters Y, the database is restored without the system-specific data. If the user enters N, the database is restored with the system-specific data and any existing system-specific data is overwritten.

Manual database backup according to rule in Element Manager

To initiate the database backup according to rule manually using Element Manager, do the following.

Manual database backup according to rule in Element Manager

1. From the Element Manager navigation tree, select **Tools > Backup and Restore > Call Server > Backup**.

   The Call Server Backup Web page opens.

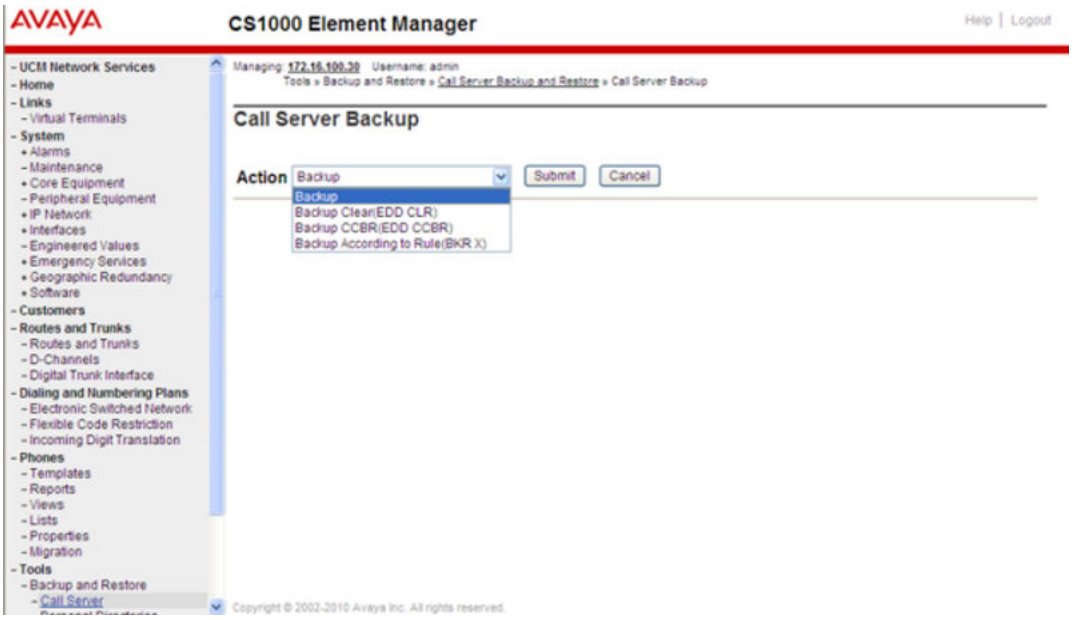

   Figure 25: Element Manager Call Server Backup Web page

2. Select **Backup According to Rule (BKR X)** from the list.

   The configured backup rule number is displayed.

3. Select the **Input Value** check box to select the backup rule number.
4. Click **Submit**.

Manual database restore in Element Manager

To initiate the database replication manually using Element Manager, do the following.

Manual database restore in Element Manager

1. From the Element Manager navigation tree, select **Tools > Backup and Restore > Call Server > Restore**.

   The Call Server Restore Web page opens.

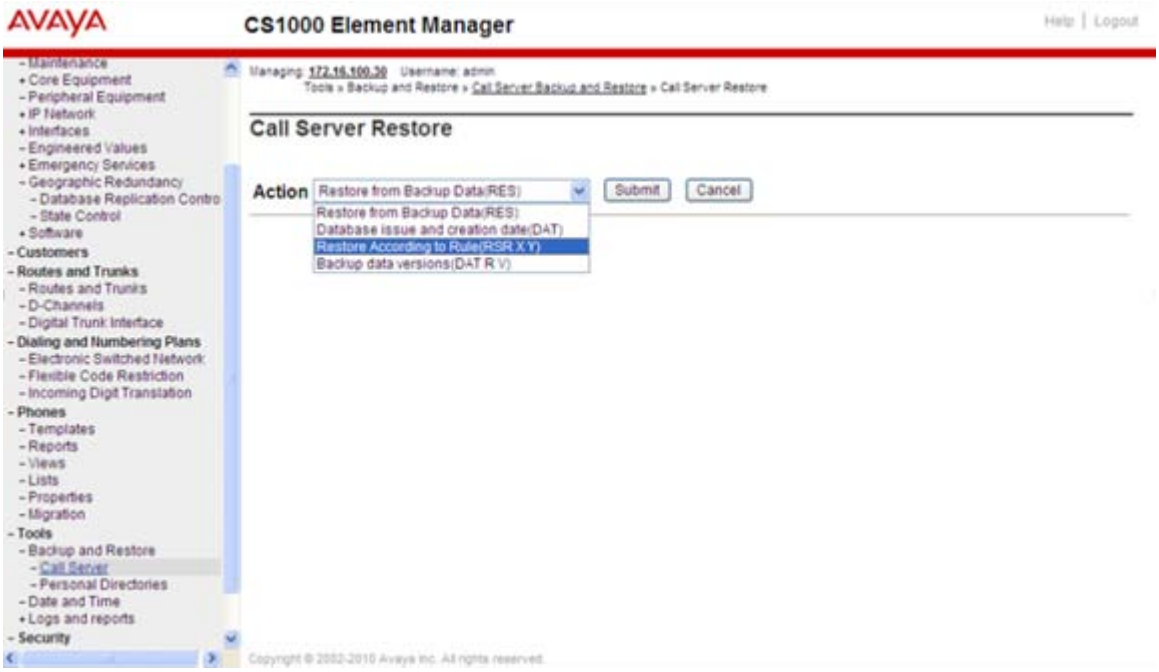

Figure 26: Element Manager Call Server Restore Web page

2. Select **Restore According to Rule** from the list.

   The configured backup rules and restore versions are displayed.

3. Select the **Apply Filtering** check box to filter system-specific data.

   This restores the file without the system-specific data that is automatically included with the compressed file sent from the Primary Call Server. If not selected, the system-specific data from the restore file is included with the restored information and overwrites any existing system-specific data on the Secondary Call Server.

   System-specific data includes IP addresses, netmasks, routes, nodes, and EPT information included with the compressed backup file sent from the Primary Call Server.

4. Click **Submit**.

Diagnostics

System status

The status of each system in the Survivable Media Gateway configuration can be obtained in LD 135. On the Primary system, the system status identifies when the last successful database replication was completed, as well as whether the replication process is operating properly.

On the Secondary system, the system status identifies the Secondary system operating state, as well as the time and date of the last successful restore and sysload operations.

Obtain the system status for Primary or Secondary systems in LD 135 as follows:

Table 13: LD 135: Survivable Media Gateway system status

Command	Description
STAT GR	The current status of Survivable Media Gateway on the specified system appears.

STAT IPL xx	Status of the IP link for a specified MG 1000E appears and indicates if it is registered to an Alternate Call Server, where:
	• xx = cabinet number of the Media Gateway queried.
	Sample output:
	`.stat ipl`
	`IPMG 4 0: LINK UP`
	`IPMG 4 1: LINK UP`
	`IPMG 12 0: LINK UP`
	`IPMG 12 1: LINK DOWN`
	`IPMG 16 0: LINK UP`
	Note:
	You can also use the STAT IPMG command in LD 117 to see more details about IP connection status, such as IPMG registration status and uptime.

SMGR xx yy	Forces Media Gateway to register on a specified Call Server, where:
	• xx = specified Call Server. Options for this parameter are PRIM (Primary), ALT1 (Alternate Call Server 1), or ALT2 (Alternate Call Server 2).
	• yy = cabinet number of the Media Gateway to be switched.
	The following warning message appears:
	`REGISTRATION SWITCH CAUSES SERVICE INTERRUPTION!`
	`Please confirm YES/(Q)`
	An MG 1000E forced to switch to a Secondary Call Server remains registered to the Secondary Call Server until it is forced to register back to the Primary Call Server with the SMGR PRIM <cabinet number> command.

Checking system status in Element Manager

To check the system status in Element Manager, do the following:

Comments? infodev@avaya.com

Checking system status in Element Manager

1. From the Element Manager navigation tree, select **System > Maintenance > Select by Overlay > LD 135—Core Common Equipment Diagnostics > Geographic Redundancy Diagnostics**.

   The Geographic Redundancy Diagnostics Web page appears.

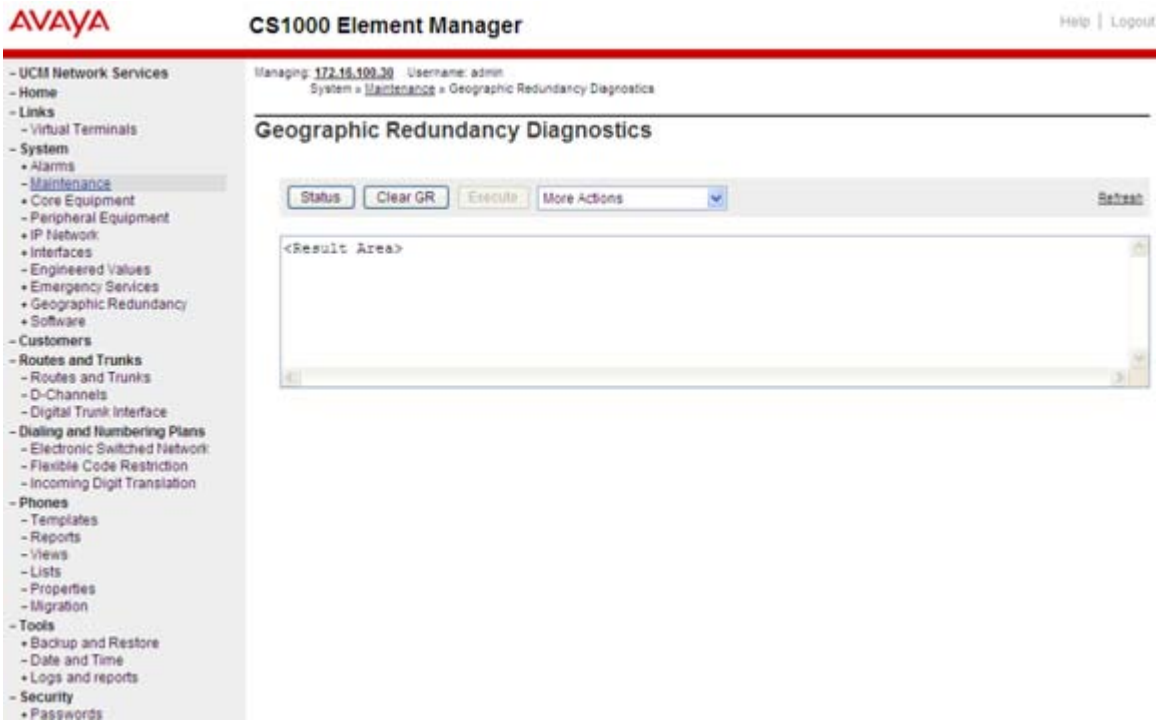

   Figure 27: Element Manager Geographic Redundancy Diagnostics Web page

2. Click **Status**.

   The system status for Survivable Media Gateway configuration displays.

   You can also check the status of individual system components.

3. Choose an item from the **More Actions** list.
4. Click **Execute**.

   The status of the selected item is displayed in the results area. If the results area displays a list (for example, of IP addresses for configured Media Gateways or available TNs), selecting the item and clicking **Execute** displays the status of the individual item.

5. To switch the registration of an item to a different Call Server, choose **Register to Primary**, **Register to Alternate CS1**, or **Register to Alternate CS2** from the **More Actions** list.

   A confirmation message appears.

6. To confirm the action, click **Ok**.

The item is registered to the new Call Server.

System faults

FTP transfer failure

During an automatic backup attempt, if the Primary system fails to gain FTP access to the Secondary system, it attempts to gain access again for a total of five attempts, with a delay of 40 seconds between attempts.

Note:

This scenario allows the Primary system to overcome short term inoperability of the Secondary system of about 200 seconds (for example, for initialization).

If a backup attempt fails, the following message appears when the Primary system status is printed (using the STAT GR command in LD 135):

```
Failed Backup attempts: x, last one at: hh:mm on MM DD, YYYY, to <SCS
IP address>
```

When the FTP transfer to the Secondary system succeeds, the counter is cleared and the `Failed Backup attempts` information no longer appears in the Primary system status.

Secondary system database endorsement failure

On the Secondary system, each database replication sysload produces the following message:

```
SRPT4645: The system will automatically be restarted for Data
Endorsement
```

If an automatic data endorsement sysload fails due to technical difficulty on the Secondary system, the following actions are automatically performed:

1. The system reverts to the last database files that were successfully loaded prior to the last restore.
2. The system performs a sysload and displays the following notification message:

   ```
   SYS0139 Database Replication Endorsement failed
   ```


Comments? infodev@avaya.com

Feature interactions

Security

The database replicated from the Primary to Secondary Call Servers is encrypted on the Primary system and decrypted on Secondary systems.

For more information on Communication Server 1000 security features, see *Security Management Fundamentals, NN43001-604*.

Integrated Call Director

Geographic Redundancy uses a defined schedule for synchronizing databases and therefore does not support the ICD (Integrated Call Director) feature. If you experience issues with database synchronization, you can perform a manual synchronization.

System monitoring

SNMP system alarms monitoring

Avaya recommends that you use Simple Network Management Protocol (SNMP) be used for system alarms monitoring. For more information about SNMP, see *Avaya Communication Server 1000 Fault Management—SNMP, NN43001-719*.

Chapter 5: Geographic Redundancy IP Phone redirection

Contents

This chapter contains information about the following topics:

Description on page 93

Normal operation on page 93

Primary system failure detection on page 97

Secondary system failure detection on page 102

NRS routing for IP Phone redirection on page 103

Upgrading an existing 1+1 configuration to Survivable Media Gateway on page 111

Description

This section contains information about IP Phone redirection for Geographic Redundancy.

Normal operation

Redirection process

In Geographic Redundancy, the redirection of IP Phones is handled differently than Survivable Media Gateway. In Geographic Redundancy mode, all IP Phones in the system are configured with their Primary Connect Server (S1) pointing to the local Secondary node Terminal Proxy Server (TPS) and their Secondary Connect Server (S2) pointing to the Primary system node TPS.

When the IP Phones are reset, they attempt to register with the Secondary system. The Secondary system accepts these registration requests and automatically attempts to redirect all IP Phones to the Primary system for registration.

IP Phones remain registered on the Secondary system only if the redirection process fails due to a failure at the Primary site or as a result of network connectivity failure.

In normal operating mode, the NRS redirects an IP Phone to the Primary system TPS for service. If the Primary Call Server is unreachable, the NRS redirects the IP Phone to Alternate Call Server 1 as the second cost route.

If both the Primary Call Server and Alternate Call Server 1 are unavailable, the IP Phone is not redirected and receives service from the local system, Alternate Call Server 2. By coordinating the IP Phone NRS routing configuration with the Survivable Media Gateway Alternate Call Servers, a geographic cluster of both IP Phones and Media Gateways can have coordinated survivability services.

Because the IP Phones and Survivable Media Gateways use a different method for redirection, there can be instances where IP Phones and SMG resources are directed to different Call Servers. This would normally only be for short periods of time (usually minutes) while the IP Phones are in transition due to failover or recovery. This can also occur for extended periods of time with partial LAN/WAN outages, meaning that IP Phone users may not be able to access users or resources in those SMGs.

During the redirection process, IP Phones display the message "server unreachable" until a connection to the server is established.

Automatic NUID

To redirect calls to the Primary system, the Secondary automatically generates a Network User ID (NUID) for each IP Phone that it registers. The NUID is similar to the Branch User ID (BUID) that is used in the Branch Office feature for the redirection of IP Phones to a main office. For more information about Branch Office, see *Avaya Branch Office Installation and Commissioning, NN43001-314*.

However, unlike the BUID in Branch Office (or the NUID in the Controlled Load-sharing configuration), which are generated manually, the NUIDs in Geographic Redundancy are generated automatically.

To automatically generate the NUID, the Secondary system uses the following formula:

Automatic NUID = (AC1 or AC2) + HLOC + DN

Table 14: Logic of NUID automatic generation on page 95 describes in greater detail the logic used by the Secondary system when it generates the NUID.

Table 14: Logic of NUID automatic generation

Variable	Logic
AC1 or AC2	In LD 15, Customer Data block, if LOC is associated with AC2, the Secondary system uses AC2 to build the NUID. Otherwise, it uses AC1. Values for AC1 or AC2 are defined in LD 86. (For more information, see *Avaya Software Input/Output Administration, NN43001-611*.)
HLOC	The Secondary system generates the NUID using the value of HLOC as defined in LD 15. (Configure the same HLOC in LD 90 for the appropriate AC1 or AC2. For more information, see *Avaya Software Input/Output Administration, NN43001-611*.)
DN	To choose the DN value for the NUID, the Secondary system scans the keys of the IP Phone, beginning from key 0 to the last key. The scan is stopped when a key with any of the following functions is met: ACD, MCN, MCR, PVN, PVR, SCN, or SCR. The DN associated with the key is used to generate the NUID. If no DN meets this criteria, the NUID cannot be generated. **Note:** For the ACD key, the ACD DN or Message Center DN is used to generate the NUID.

To generate the NUID, the Secondary system uses values originally defined on the Primary system and copied to the Secondary system during the database replication process. The NUID is created with the HLOC value originally defined on the Primary system. The Secondary system can use the NUID value to query the Network Redirect Service (NRS) and determine the location of the Primary system.

Note:

To ensure that the IP Phone redirection is successful, the HLOC value for the Primary system endpoint must be defined on the NRS as the least-cost route. See Numbering plan on page 125 for details.

The following summarizes the IP Phone redirection process from the Secondary system to the Primary system:

1. An IP Phone resets and, using its S1 value, registers first with the Secondary system node TPS, and then with the Secondary system.
2. The Secondary system accepts the registration and automatically generates the NUID for the registered IP Phone.
3. The Secondary system TPS queries the NRS for the home system node indicated by the generated NUID.

4. The NRS responds with the least-cost route for the HLOC of the NUID. (The Primary system HLOC must be configured as the least-cost route.)
5. When the Secondary system receives the positive response, it redirects the IP Phone to the Primary system.
6. The redirected IP Phone registers to the Primary system.

Note:

Geographic Redundancy uses the NUID for redirection only. When the IP Phone registers on the Primary system, the phone uses the DN values that are defined on the Primary system.

Note:

Due to the database replication process, the TN of the IP Phone is the same on both Primary and Secondary systems. Therefore, a Network Home TN (NHTN) need not be defined in LD 11 for the IP Phones in a Geographic Redundancy configuration. For more information about NHTN, see Geographic Redundancy Controlled Load-sharing configuration on page 115.

Figure 28: IP Phone redirection: Normal operation on page 97 shows the normal IP Phone registration attempt and redirection process.

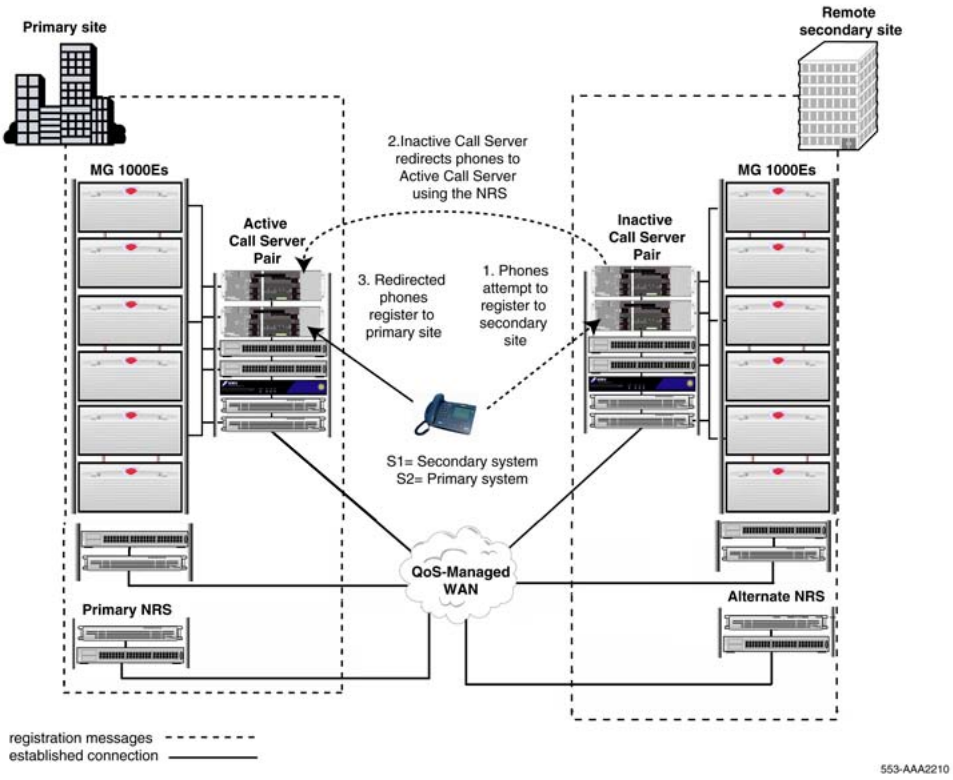

Figure 28: IP Phone redirection: Normal operation

When both Primary and Secondary systems remain operational, all IP Phones are redirected by the Secondary system and register with the Primary system. The Primary system provides service to all IP Phones.

The Secondary system redirection also provides a practical means to monitor the network connections between the IP Phones, Primary system, and Secondary system.

While S1 points to the Secondary system, all IP Phones must have their local secondary Connect Server (S2) pointing to the Primary system. This ensures that the IP Phones can register directly to the Primary system and continue normal operations if the connection to the Secondary system is lost.

Primary system failure detection

The transfer of system control from a failed Primary system to the Secondary system is not a traditional redundant Call Server switchover operation.

If the Secondary system cannot redirect the IP Phones to the Primary system because of a Primary system failure (or network connectivity problem), the IP Phones stay registered on the Secondary system. Therefore, each additional IP Phone that remains registered on the Secondary system represents a potential problem with the Primary system.

Note:

Failure of the IP Phone to register to the Primary system is not necessarily caused by a failure of the Primary system. The cause can also be LAN/WAN connectivity problems. Geographic Redundancy does not differentiate between these types of failures.

To monitor Primary system health, the Secondary system maintains a real-time count of IP Phones (N) registered on the Secondary system. Once the number of IP Phones (N) registered on the Secondary system exceeds the Geographic Redundancy Threshold (GRTHR), the Secondary system escalates to ACTIVATING state. (The customer can define GRTHR in LD 117.)

Figure 29: IP Phone redirection: Primary site failure on page 98 shows a primary site failure.

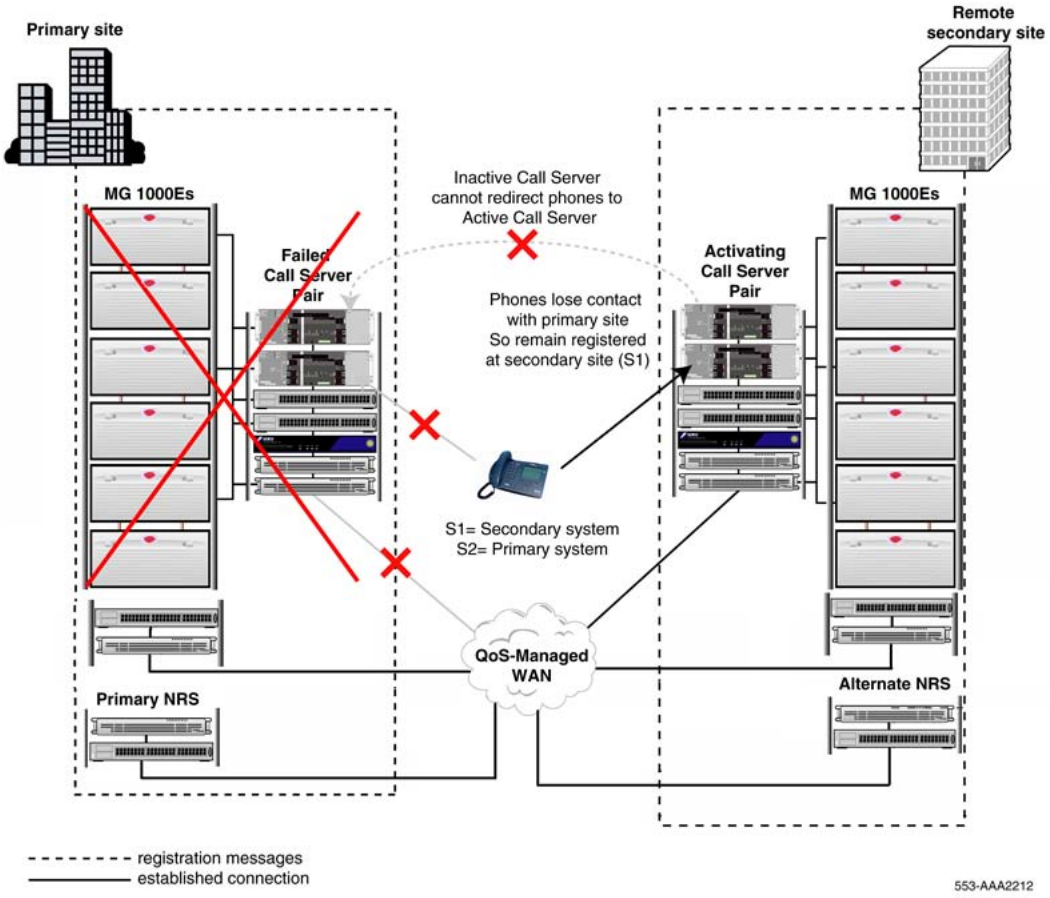

Figure 29: IP Phone redirection: Primary site failure

While in the ACTIVATING state, the Secondary system provides the IP Phones with necessary service, but does not become fully active. This allows the Secondary system to take into account a short-term failure at the primary site.

If the Primary system regains full operation within a short period of time, as defined by the Short Term Failure Timer (STFT), the Secondary system redirects all IP Phones to the Primary system, and normal operation resumes.

Only when the Primary system remains out-of-service past the STFT does the Secondary system escalate to the ACTIVE state. This ensures that the ACTIVE state operating license period is not used unnecessarily.

Note:

The Secondary system ACTIVE state is limited by an operating license period of 90 days.

Secondary system operating states

The Secondary system provides a number of operating states that allow it to transition smoothly from INACTIVE to ACTIVE and back again. Refer to Figure 30: Secondary system operating state logic on page 100 and Table 15: Secondary system state control data set on page 101 for a description of the Secondary system state logic and the data set, including timers, that control the transitions between the various Secondary system states.

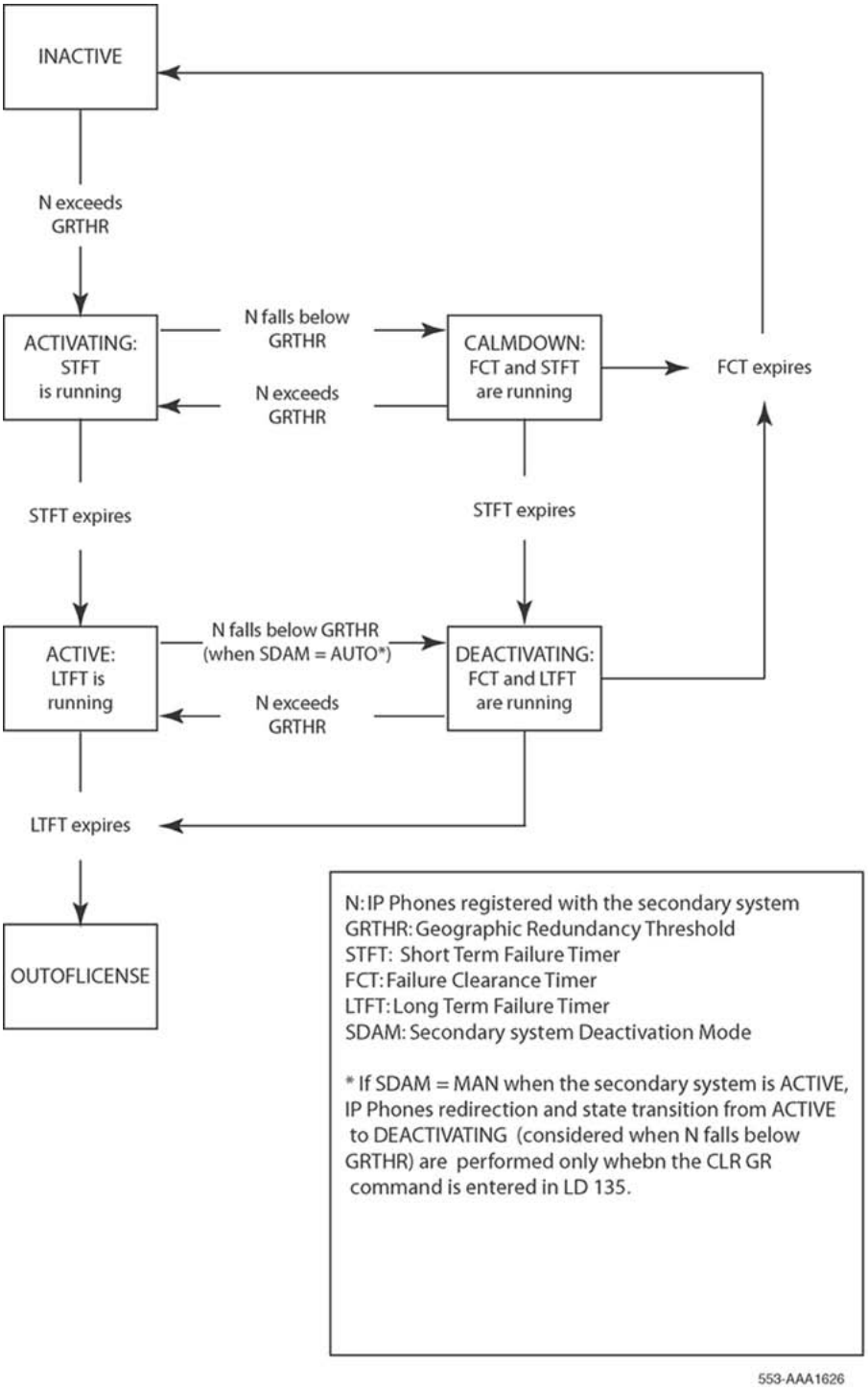

Figure 30: Secondary system operating state logic

Table 15: Secondary system state control data set

Variable	Definition
N	Number of IP Phones registered with the Secondary system, updated in real time. The Secondary system uses this value to monitor Primary system health.
GRTHR	Geographic Redundancy Threshold: a customer-definable threshold against which N is compared. If the Secondary system is INACTIVE and N exceeds the GRTHR, the Secondary system escalates to the ACTIVATING state. (GRTHR is configurable in LD 117.)
STFT	Short Term Failure Timer: the short period of time during which the Primary system can remain inoperable without the Secondary system assuming full system control (for example, as caused by initialization, sysload, or minor technical failure). STFT starts when the Secondary system enters the ACTIVATING state (N exceeds GRTHR). If the STFT expires while the system is in the ACTIVATING state, the Secondary system escalates to the ACTIVE state. (STFT is configurable in LD 117.)
FCT	Failure Clearance Timer: period of time that must elapse once the Primary system is brought back online (and N falls below GRTHR) before the Secondary system state can revert to INACTIVE. This timer serves to prevent unnecessary transitions between less severe and more severe states when the value of N is close to GRTHR. (FCT is configurable in LD 117.) **Note:** FCT is also used with the CLR GR command in LD 135. FCT defines the period of time during which redirection attempts to the Primary system are allowed.
LTFT	Long Term Failure Timer: license period during which the Secondary system is allowed to run in the ACTIVE state. (LTFT is a hardcoded value of 90 days. It is not a customer-configurable value.) When LTFT expires, the system enters the OUTOFLICENSE state.
SDAM	Secondary system Deactivation Mode: specifies whether the Secondary system is in automatic deactivation (AUTO) or manual deactivation (MAN) mode. If SDAM = AUTO, the Secondary system, once ACTIVE, queries the NRS for the Primary system address every ten minutes. When the Primary system comes back online the Secondary system can resume the redirection of IP Phones to the Primary system and automatically revert to the DEACTIVATING state (when N falls below GRTHR). If SDAM = MAN, the Secondary system, once ACTIVE, stops trying to redirect IP Phones to the Primary system. It stays in the ACTIVE state until the CLR GR command is initiated manually in LD 135. The CLR GR command triggers the redirection attempts to the Primary system to resume (for a maximum period defined by FCT). If the redirections are successful

Variable	Definition
	and N falls below GRTHR, the Secondary system reverts to the DEACTIVATING state. (SDAM is configurable in LD 117)

For details on configuring the Geographic Redundancy State Control Block, refer to Configuring GRSC block in Element Manager on page 71.

For details on the recovery of the Primary system following a Long-Term Failure, refer to Primary system recovery on page 82.

Secondary system operating state survival

The Secondary system operating state can survive an initialization or a sysload (the operating state is restored from non-volatile memory). However, if the operating state is restored to ACTIVE after a sysload or initialization, the state transition from ACTIVE to DEACTIVATING is prevented for a period of 15 minutes. This waiting period eliminates unnecessary state transitions as it allows the IP Phones to re-register to the Secondary system. The Secondary system can then obtain the appropriate value of IP Phones registered (N).

OUTOFLICENSE state

When the Secondary system remains in the active state beyond the LTFT of 90 days and enters the OUTOFLICENSE state, some functional limitations are imposed on the Secondary system. `Beyond licensed period` displays on all IP Phones and `Licensed period is exceeded` appears on the TTY banner upon successful login.

As well, all the data dump (midnight and manual), automatic restore, and automatic sysload operations are restricted.

To clear the OUTOFLICENSE state, a software installation is required.

Secondary system failure detection

Figure 31: IP Phone redirection: Secondary site failure on page 103 shows a secondary site failure.

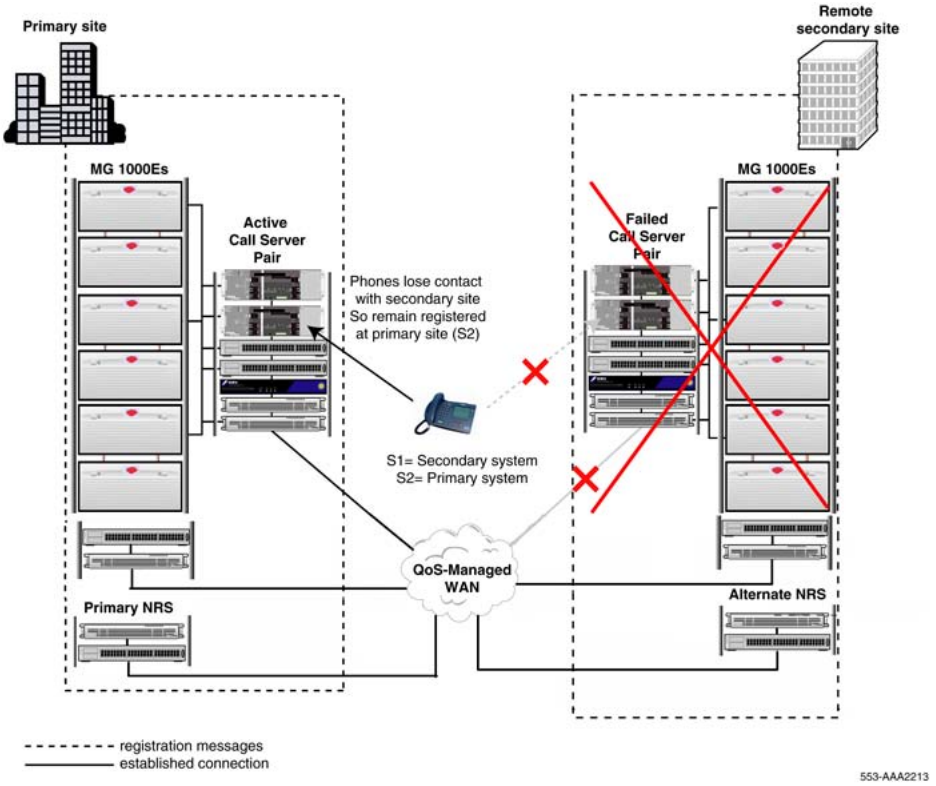

Figure 31: IP Phone redirection: Secondary site failure

If the Secondary system fails while the IP Phones are registered on the Primary system, the Primary system continues to provide normal telephone service. If an IP Phone is reset, the IP Phone attempts to connect to the Secondary system a number of times, defined by the S1 Retry Count of the IP Phone. Once its S1 Retry Count has expired, the IP Phone uses its S2 value to register directly to the Primary system and receive service.

To ensure that Geographic Redundancy is available when required, the Secondary system fault must be corrected and the system restored.

NRS routing for IP Phone redirection

The following information is provided in addition to the normal configuration of NRS as described in *Avaya Network Routing Service Fundamentals, NN43001-130*.

IP Phones use a different mechanism than the Survivable Media Gateway for locating the Alternate Call Servers in the Geographic Redundancy configuration. They use a similar method to Branch Office (BO) for IP Phone redirection. During the redirection process, the terminal proxy server that controls the IP Phone will reference the Network Routing Server (NRS) to locate the Primary Call Server or Alternate Call Servers for the IP Phone to register to. The selection is made by the least cost route currently configured and registered on the NRS.

To properly redirect IP Phones in the event of failure, the **Network Connection Server enabled** box must be selected in the Gateway Endpoints section of the NRS Manager Configuration page of the signaling server, as shown in Figure 32: NRS Configuration with Network Connection Server enabled on page 104. A Secondary Call Server can register as a either an H.323 or SIP endpoint and still have Network Connection Server (NCS) services available.

Important:

If you do not enable Network Connection Server, Geographic Redundancy will not function properly.

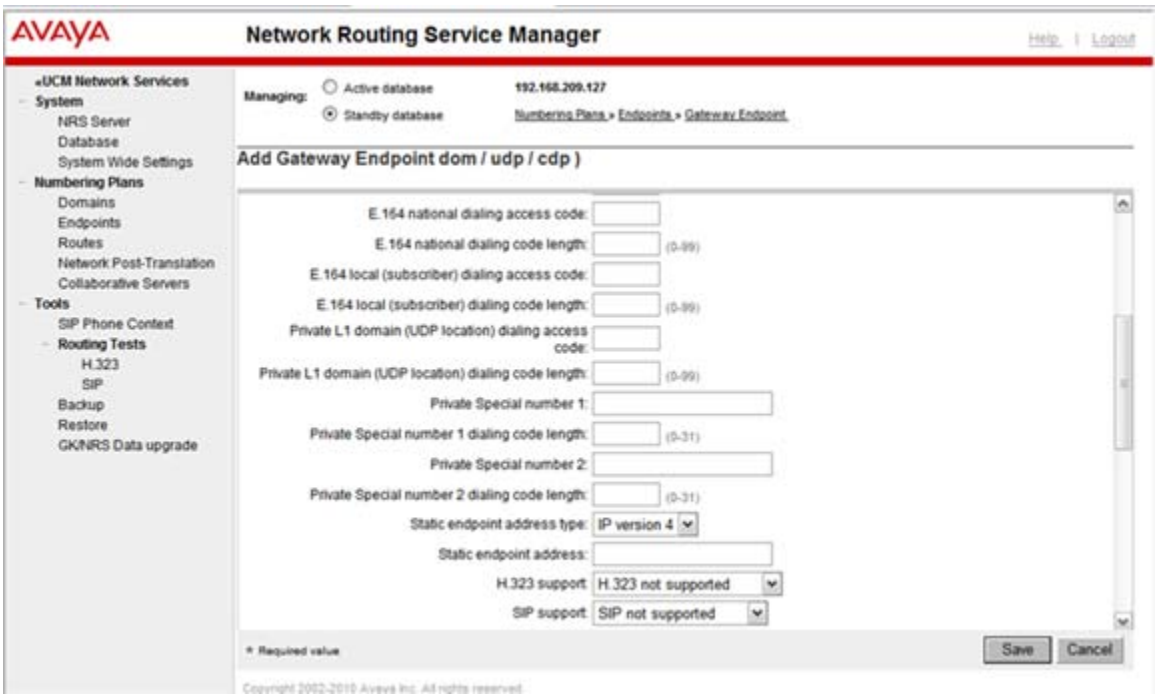

Figure 32: NRS Configuration with Network Connection Server enabled

Node IDs

All Secondary Call Servers use the same node ID as the Primary Call Server. To prevent multiple leaders having the same ID, the Signaling Servers in the Geographic Redundancy configuration must be on different TLAN subnets.

Configuration examples

Geographic Redundancy is very flexible in how the IP Phones utilize the Secondary Call Servers and there are many different configuration options available. For example, there could be a single Secondary Call Server for all IP Phones, there could be multiple Secondary Call

Servers with one at each physical location, or there could be Secondary Call Servers supporting geographic clusters of sites best suiting the redundancy requirements.

There are four basic configurations:

- Simple redundancy on page 105
- Multiple site redundancy on page 106
- Triple redundancy on page 107
- Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway on page 109

Simple redundancy

Simple redundancy consists of a Primary Call Server and a Secondary Call Server, as shown in Figure 33: Simple redundancy on page 105.

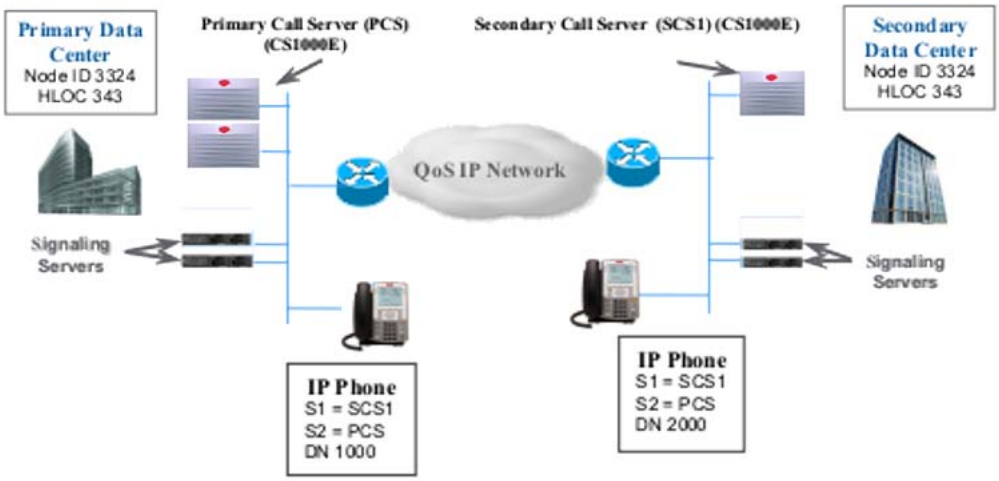

Figure 33: Simple redundancy

In this example, both IP Phones are configured with their Signaling Server 1 (S1) routing to Secondary Call Server 1 (SCS1). Signaling Server 2 (S2) is routing to the Primary Call Server as backup redirection in the event the secondary site is out of service. The routing cost for the primary location is always set to 1 (lowest cost) to ensure that all IP Phones are redirected to the Primary Call Server. The Secondary Call Server (SCS) has a cost factor of 2, being the second choice.

Table 16: NRS routing entries for simple redundancy on page 106 shows the values used in this example to configure the NRS routing entries for the IP Phones. The NRS directs all calls and IP Phones to the Primary Call Server while the Primary Call Server is registered to it; if the Primary Call Server is not reachable or is inactive, the NRS will return the second cost route and redirect calls and IP Phones to the Secondary Call Server.

Table 16: NRS routing entries for simple redundancy

UDP and CDP	Call Server
UDP 343	cost 1 Primary Call Server
UDP 343	cost 2 Secondary Call Server
CDP 10	cost 1 Primary Call Server
CDP 10	cost 2 Secondary Call Server
CDP 20	cost 1 Primary Call Server
CDP 20	cost 2 Secondary Call Server

Multiple site redundancy

Geographic Redundancy is flexible in how the IP Phones utilize the Secondary Call Servers. Depending on the redundancy requirements, there could be a single Secondary Call Server for all IP Phones, multiple Secondary Call Servers with one at each physical location, or Secondary Call Servers supporting geographic clusters of sites.

Figure 34: Multiple site redundancy on page 106 shows a Primary Call Server (PCS) and multiple Secondary Call Servers (SCS).

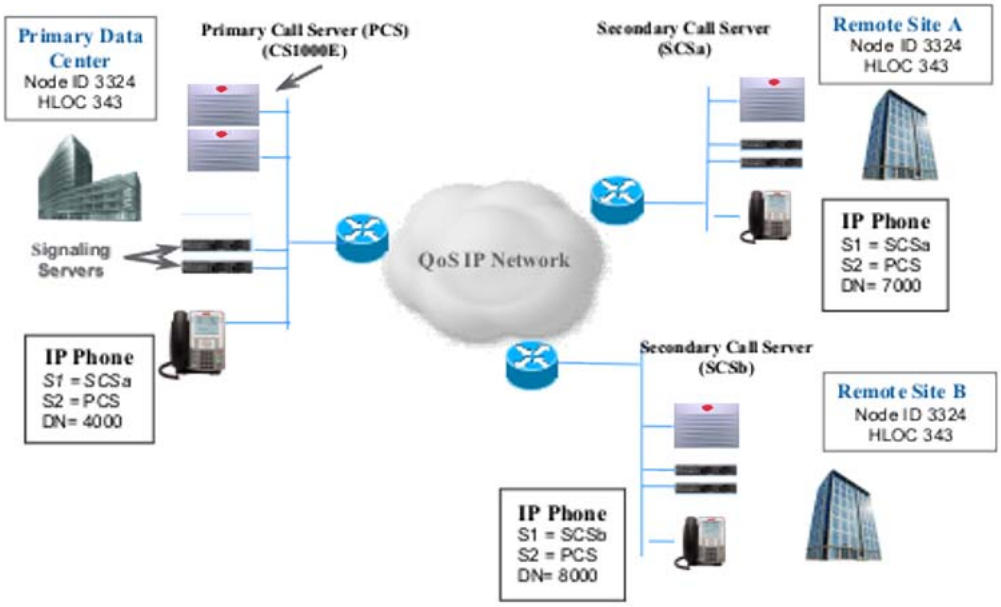

Figure 34: Multiple site redundancy

In this diagram, remote IP Phones have S1 pointing to the Secondary Call Server (SCSx) at their own geographical location and S2 pointing to the Primary Call Server. The IP Phones located at the Primary site can also have their S1 pointing to one of the Secondary sites. This

is one way of configuring local redundancy at each remote location to provide protection against WAN failure.

The routing cost for the Primary location is always set to 1 (lowest cost) to ensure that all IP Phones are redirected to the Primary Call Server. The Secondary Call Servers have a cost factor of 2, being the second choice.

See Table 17: NRS routing entries for multiple site redundancy on page 107 to configure the NRS routing entries for the IP Phones for the above example. In all cases, the NRS will redirect the calls and IP Phones to the Primary Call Server as long as it is registered to the NRS. If the Primary Call Server is inaccessible, the NRS will return the second cost route and direct calls and IP Phones to the Secondary Call Server.

Table 17: NRS routing entries for multiple site redundancy

UDP and CDP	Call Server
UDP 34370	cost 1 Primary Call Server
UDP 34370	cost 2 Secondary Call Server (SITE A)
CDP 70	cost 1 Primary Call Server
CDP 70	cost 2 Secondary Call Server (SITE A)
UDP 34380	cost 1 Primary Call Server
UDP 34380	cost 2 Secondary Call Server (SITE B)
CDP 80	cost 1 Primary Call Server
CDP 80	cost 2 Secondary Call Server (SITE B)
UDP 34340	cost 1 Primary Call Server
UDP 34340	cost 2 Secondary Call Server (SITE A)
CDP 40	cost 1 Primary Call Server
CDP 40	cost 2 Secondary Call Server (SITE A)

Triple redundancy

In this configuration, each IP Phone has 3 different Call Servers from which it can receive service. This configuration can be aligned with the triple redundancy available with Survivable Media Gateway.

Triple redundancy configurations can be applied in networks having Primary and Secondary data centers with multiple remote locations. Under normal operating conditions, the IP Phones direct their S1 to the local Secondary Call Server 2 and the NRS routes the redirection to the Primary Call Server site as first choice and the Secondary Call Server 1 site as second choice.

Figure 35: Triple redundancy on page 108 shows a triple redundancy configuration:

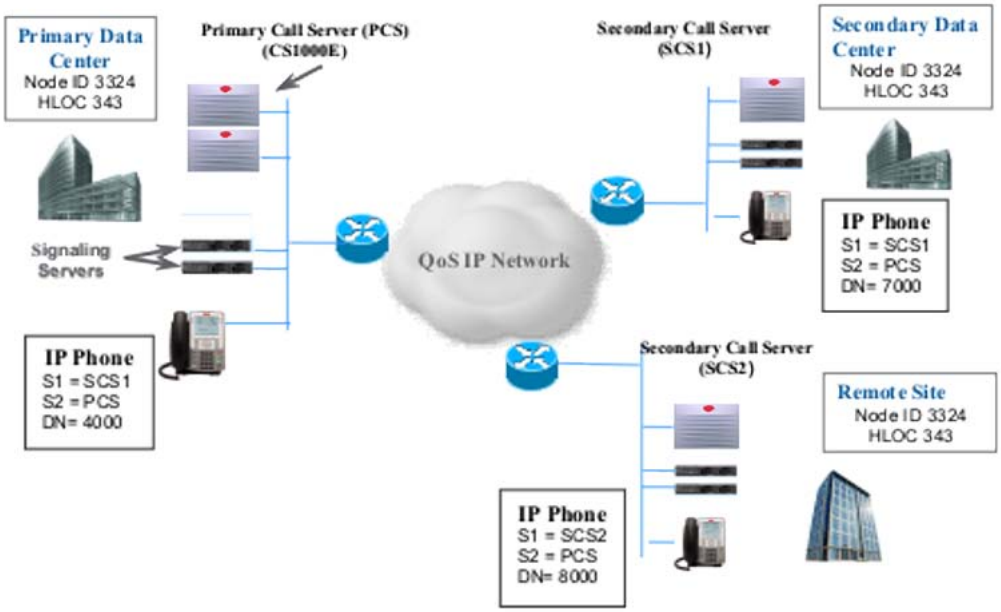

Figure 35: Triple redundancy

SeeTable 18: NRS routing entries for triple redundancy on page 108 to configure the NRS routing entries for the IP Phones in the example above. The NRS redirects the calls and phones to the Signaling Server Gateway at the Primary Call Server for as long as the Primary Call Server is registered to the NRS. If the Primary call server is inaccessible, the NRS returns the second cost route and directs calls and IP Phones to the Secondary Call Server (SCS1). If both PCS and SCS1 are inaccessible, the IP Phones register to the local call server (SCS2).

Table 18: NRS routing entries for triple redundancy

UDP and CDP	Call Server
UDP 34340	cost 1 Primary Call Server
UDP 34340	cost 2 Secondary Call Server 1
CDP 40	cost 1 Primary Call Server
CDP 40	cost 2 Secondary Call Server 1
UDP 34370	cost 1 Primary Call Server
UDP 34370	cost 2 Secondary Call Server 1
CDP 70	cost 1 Primary Call Server
CDP 70	cost 2 Secondary Call Server 1
UDP 34380	cost 1 Primary Call Server
UDP 34380	cost 2 Secondary Call Server 1
UDP 34380	cost 3 Secondary Call Server 2

UDP and CDP	Call Server
CDP 80	cost 1 Primary Call Server
CDP 80	cost 2 Secondary Call Server 1
CDP 80	cost 3 Secondary Call Server 2

This example shows a more complex arrangement of both a Primary (PCS) and Secondary (SCS1) Call Server in addition to having local survivability with a local Call Server (SCS2).

Having the Secondary (SCS1) as a central data centre allows all IP Phones to failover to the same Call Server if the Primary Call Server fails. This maintains the system as a homogeneous single system. If the Secondary (SCS1) also fails or there is a WAN outage, the remote IP Phones can redirect to their own local Call Server (SCS2), but the system will have divided into multiple individual systems.

IP Phones located at the Primary site can also have their S1 pointing to the Secondary data centre (SCS1), providing additional redundancy.

The routing cost for the Primary location is always set to 1 (lowest cost) to ensure that all IP Phones get redirected to the Primary Call Server. The Secondary Call Server (SCS1) at the backup data centre will have a cost factor of 2, being the second choice. The local Secondary Call Servers (SCS2) will have a cost factor of 3, as the third choice.

Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway

Triple redundancy can be extended to include Survivable Branch Office (SBO) or Survivable Remote Gateway (SRG) at the remote site.

Using this layered approach provides up to 4 levels of survivability to Branch IP Phone users. This configuration is most likely not required as SBO and SRG already provide their own level of redundancy and would normally only be directly tied to the Primary site.

Figure 36: Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway on page 110 shows an example of this type of configuration.

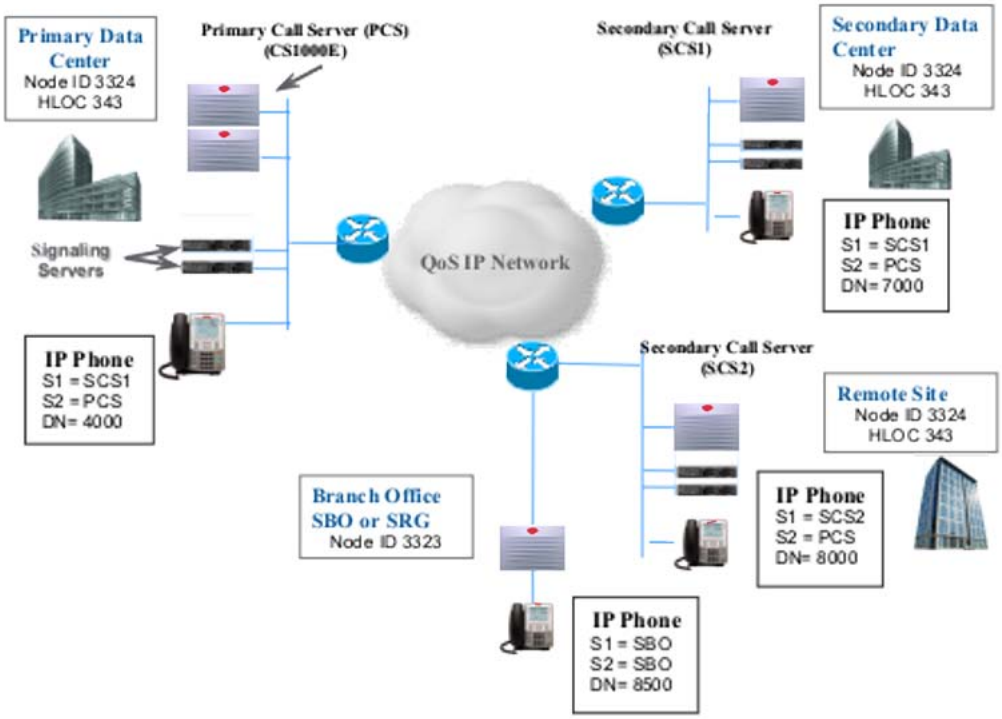

Figure 36: Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway

Table 19: NRS routing entries for Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway on page 110 shows only the extra entries required for the Survivable Branch Office. To ensure that all IP Phones get redirected to the Primary Call Server, the routing cost for the Primary location is always set to 1 (lowest cost). The Secondary Call Server (SCS1) at the backup data centre will have a cost factor of 2, as the second choice. The local Secondary Call Servers (SCS2) have a cost factor of 3, as the third choice. The Branch Office is configured with a route cost of 4, as the fourth choice.

Table 19: NRS routing entries for Geographic Redundancy with Survivable Branch Office or Survivable Remote Gateway

UDP and CDP	Call Server
CDP 85	cost 1 Primary Call Server
CDP 85	cost 2 Secondary Call Server 1
CDP 85	cost 3 Secondary Call Server 2
CDP 85	cost 4 Branch Office

Upgrading an existing 1+1 configuration to Survivable Media Gateway

This section describes how to upgrade an existing Geographic Redundancy 1+1 configuration to a Geographic Redundancy configuration with Survivable Media Gateway. This feature requires Avaya Communication Server 1000 Release 5.0 or later to be installed on the Primary and all Secondary Call Servers. Survivable Media Gateway is not supported for previous versions of the software and is only supported on Avaya CS 1000E (CP PIV and CP PM) Call Servers.

For information on system upgrades, see the following documents, as applicable:

- *Avaya Communication Server 1000E Upgrade—Option 11C Cabinet to Avaya CS 1000E, NN43041-464*
- *Avaya Communication Server 1000E Upgrade—Option 11C Chassis to Avaya CS 1000E, NN43041-465*
- *Avaya Communication Server 1000E Upgrade—Avaya CS 1000M Cabinet to Avaya CS 1000E, NN43041-466*
- *Avaya Communication Server 1000E Upgrade—Avaya CS 1000M Chassis to Avaya CS 1000E, NN43041-467*

To upgrade an existing 1+1 configuration to Survivable Media Gateway, you must complete the following tasks:

- Upgrade the Primary Call Server and all Secondary Call Server to Communication Server 1000 Release 5.0 or higher. For more information on upgrading system software, see *Avaya Communication Server 1000E Upgrades, NN43041-458*.
- On the Primary Call Server, update the Geographic Redundancy configuration for the existing Secondary Call Server.
- On the Primary Call Server, update the Geographic Redundancy configuration for the new Secondary Call Server.
- Replicate the database from the Primary Call Server to each Secondary Call Server.
- On each Secondary Call Server, verify and accept the transferred database.

These tasks are described in more detail in Upgrading an existing 1+1 configuration to Geographic Redundancy with Survivable Media Gateway on page 112.

Upgrading an existing 1+1 configuration to Geographic Redundancy with Survivable Media Gateway

1. Upgrade the Primary Call Server to Avaya Communication Server 1000 Release 5.0 or higher. The system might indicate that one or all of the following warning conditions exist:
   - Several Secondary Call Server backup rules are configured for a server with the same IP address.
   - Several RMD backup rules are configured.
   - Several FMD backup rules are configured.
2. On the new Secondary Call Server:
   - Install the hardware.
   - Install Avaya Communication Server 1000 Release 5.0 or higher software.
   - Configure the Call Server. For information on configuring a Call Server, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.
3. On the Primary Call Server:
   - Configure the new peripheral equipment.
   - In LD 117, update the Geographic Redundancy configuration for the existing Secondary Call Server by doing the following:
     - Select the RMD, FMD, and SCS backup rule to use and remove the redundant ones.
     - Configure the BKPS block associated with the SCS backup rule. For more information, see Configure a Backup Schedule (BKPS) on page 61.
     - Modify the GRDRC block and configure the ABKUP field to SCHD. For more information, see Configure Database Replication Control (GRDRC) block on page 65.
     - Modify the GRSC block to associate it with the existing Secondary Call Server and configure the proper parameters. For more information, see Configure State Control (GRSC) Blocks on page 70.
   - In LD 117, configure the Geographic Redundancy for the new Secondary Call Server by doing the following:
     - Configure the backup rule associated with the new Secondary Call Server. For more information, see Configure a Backup Rule (BKR) on page 58.
     - Configure the BKPS block associated with the new Secondary Call Server. For more information, see Configure a Backup Schedule (BKPS) on page 61.
     - Configure the GRSC block associated with the new Secondary Call Server. For more information, see Configure State Control (GRSC) Blocks on page 70.

- In LD 43, use the BKR command with the appropriate backup rule to replicate the database to both the old and new Secondary Call Servers. For more information, see Manual database replication and restore on page 85.

4. On the new Secondary Call Server:
   - Modify the GRDRC block by configuring the ABKUP field to SCHD. For more information, see Configure Database Replication Control (GRDRC) block on page 65.
   - Verify that the backup set from the Primary Call Server is received.
   - Define Backup Rule 1. For more information, see Configure a Backup Rule (BKR) on page 58.
   - Use the RSR 1 command in LD 43 to manually restore the backup data. It is mandatory to use filtering.
   - Sysload the Call Server to accept the database replication.

Chapter 6: Geographic Redundancy Controlled Load-sharing configuration

Contents

This section contains information on the following topics:

Description on page 115

Normal operation on page 116

Site 1 system failure on page 118

Site 2 system failure on page 120

Planning a Controlled Load-sharing configuration on page 121

Numbering plan on page 125

Branch Office support on page 126

Installing a Controlled Load-sharing configuration on page 129

Maintenance on page 131

Feature interactions on page 131

System monitoring on page 132

Description

The Controlled Load-sharing configuration increases the reliability of Avaya Communication Server 1000M (Avaya CS 1000M) systems (CP PIV) and Avaya CS 1000E systems through configuration of two geographically-separated systems that provide redundancy for each other. In this configuration, both systems are active and perform call processing for their local telephones. If either system fails, the remaining active system can assume control of the IP Phones on the failed system and provide service as normal.

The two systems in a Controlled Load-sharing configuration do not have to be of the same type.Avaya CS 1000M Systems and Avaya CS 1000E systems provide IP Phone Redundancy for either an Avaya CS 1000M System or Avaya CS 1000E system.

Note:

The Controlled Load-sharing configuration provides redundancy for IP Phones only. Redundancy for analog (500/2500-type) telephones and digital telephones is not supported. Analog (500/2500-type) and digital telephones can still be connected to a system, but they are not operational if that system fails.

Normal operation

Figure 37: Controlled Load-Sharing configuration on page 116 shows two Avaya CS 1000E systems in a Controlled Load-Sharing configuration.

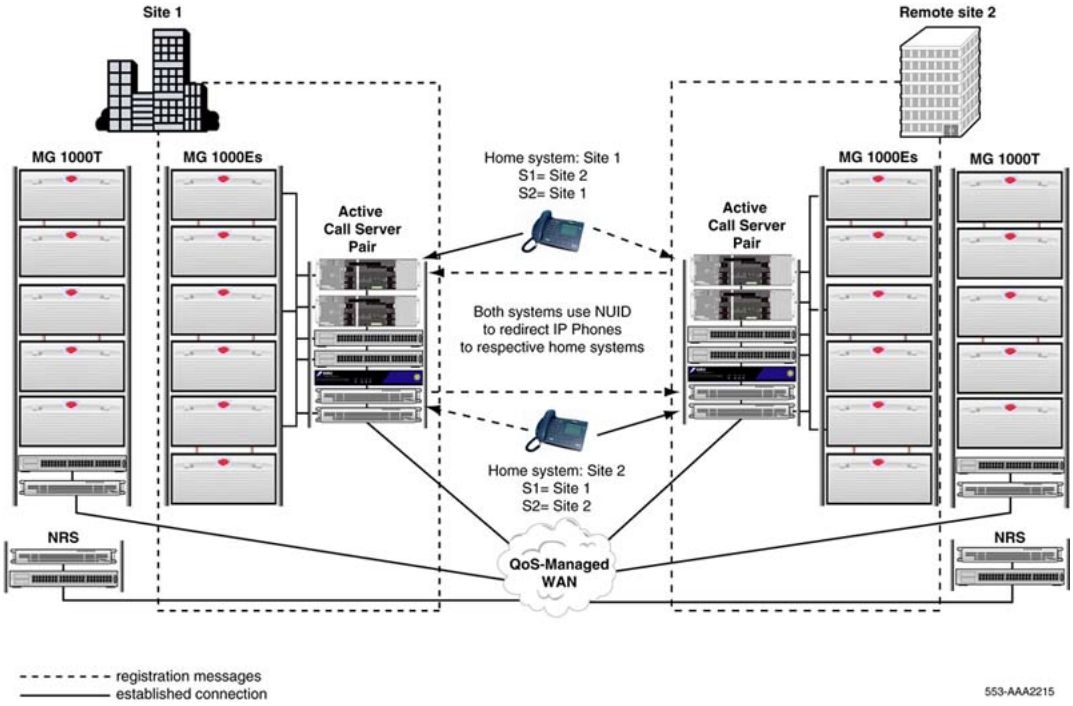

Figure 37: Controlled Load-Sharing configuration

Connectivity between the two systems is provided through a QoS-managed WAN.

The Controlled Load-sharing configuration employs similar functionality to the Branch Office feature. For each IP Phone, a DN and TN must be configured on a home system, where the IP Phone ultimately registers. At the backup system, the IP Phone is also assigned a DN and TN. This ensures that the backup system can provide the necessary functionality in case of failure at the home system.

Note:

The DNs and TNs configured on the home and backup systems must be duplicates.

On the backup system, each IP Phone is also manually assigned a Network User ID (NUID) and Network Home TN (NHTN), similar to the Branch User ID (BUID) and Main Office TN (MOTN). The NHTN corresponds to a home system TN; the NUID corresponds to a dialable home system DN where the IP Phone ultimately registers. As a result, the backup system can use the NUID value to redirect IP Phones to the home system for registration.

Redirection process

In the Controlled Load-sharing configuration, each redundant IP Phone's Primary Connect Server (S1) points to its backup system. When the IP Phone powers up, it registers first with the backup system node Terminal Proxy Server (TPS), and then with the backup system itself. The backup system, reading the IP Phone's NUID, automatically redirects the IP Phone to the home system — the backup TPS queries the NRS for the IP address of the home system node indicated by the NUID. When it receives a positive response, the IP Phone is redirected to the home system.

IP Phones remain registered at the backup system only if the connection to the home system is lost.

While each IP Phone's S1 points to its backup system, its Secondary Connect Server (S2) must point to its home system. This ensures that the IP Phones can register directly to their home system and continue normal operations if the connection to their backup system is lost.

In Figure 37: Controlled Load-Sharing configuration on page 116, both systems serve as home and backup systems: each system provides service to IP Phones that are registered locally and redirects IP Phones to their home systems. As a result, the two systems back each other up.

Database configuration

Unlike the Geographic Redundancy with Survivable Media Gateway configuration, the Controlled Load-sharing configuration provides no database replication. The databases must therefore be configured manually at each site.

The Controlled Load-sharing configuration provides additional flexibility in the possible hardware and software configurations at either site as the systems are not constrained by the necessity to duplicate hardware and software. Instead, systems are only limited by the need to provide the required redundant capacity and services for the additional IP Phones.

To simplify database administration, the TN range used for the redundant IP Phones must be the same on the home and backup systems. As well, configure the IP Phones with the same DN, TN, and features at both sites for feature and application consistency.

As the Avaya CS 1000E systems and Avaya CS 1000M Systems use the same TN mapping format (l s c u), there is no conflict when a backup system is of a different type than the home system.

In addition, there are no limitations on installing IP Phones, analog (500/2500-type) telephones, or digital telephones that have no redundancy requirements on either system.

Software

To implement the Controlled Load-sharing configuration, there are no additional software packages required on either system. The software packages required for the Geographic Redundancy Survival Media Gateway and 1+1 configurations, 404 (GRPRIM) and 405 (GRSEC), do not apply to the Controlled Load-sharing configuration.

However, the two systems must run CS 1000 Release 4.0 (or later) software. Software installation, including installation of patches, is performed separately for each system.

Ideally, the software packages on the backup system must provide the same functionality as those offered on the home system. However, this is not a necessity if less functionality is acceptable at the backup system. Regardless, ensure that the backup system has sufficient capacity and User licenses available to provide redundancy for the additional IP Phones.

Hardware

With the Controlled Load-sharing configuration, there is greater flexibility available in configuring the hardware at both sites. The only physical constraint on a backup system is to have sufficient hardware to provide the capacity and the appropriate range of matching TNs required to service the additional redundant IP Phones.

The backup system can be configured to provide the same basic features and services as the home system. This is not a requirement if fewer services on the backup system is acceptable.

Carefully plan the backup system to provide the necessary redundant capacity required. For further planning details, see Planning a Controlled Load-sharing configuration on page 121.

Site 1 system failure

Figure 38: Load-sharing configuration: system failure at Site 1 on page 119 shows a Controlled Load-sharing configuration that is experiencing a failure at Site 1.

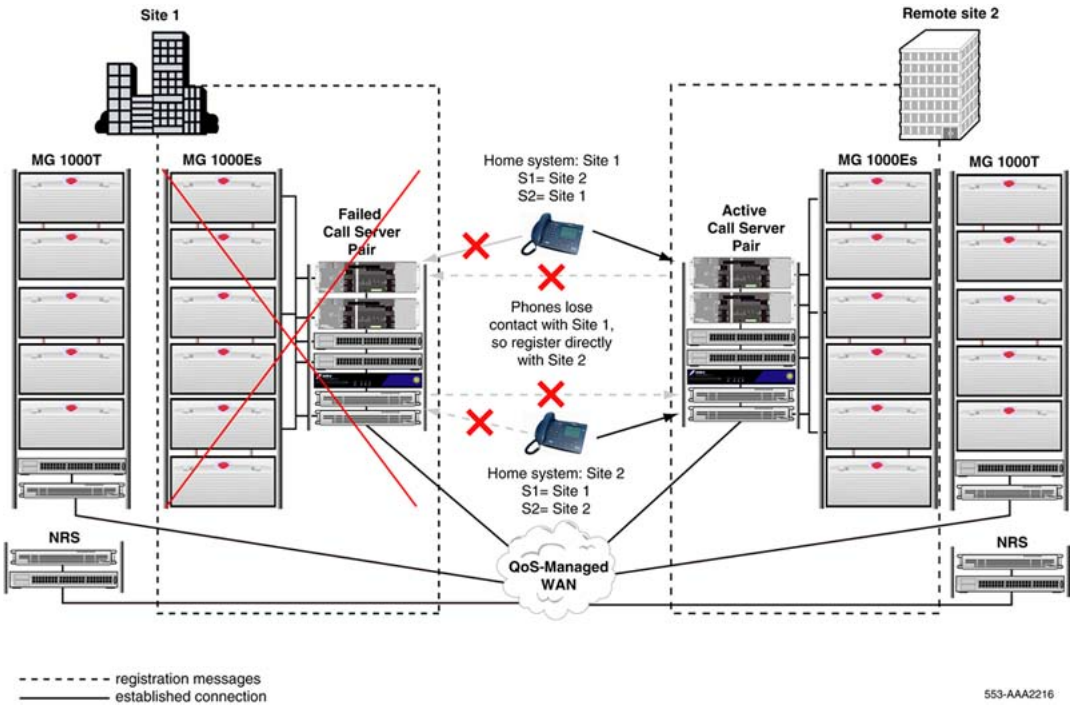

Figure 38: Load-sharing configuration: system failure at Site 1

If the IP Phones in the network cannot connect to Site 1 because of a system failure (or network connectivity problem), the IP Phones act as follows:

- Site 1 = Home system: If Site 1 is their home system (S2), the IP Phones stay registered with the Site 1 TPS for ten minutes. At the end of ten minutes, the IP Phones reboot and register first to the backup system TPS at Site 2, as normal (if the Release key is pressed on an IP Phone within ten minutes, it resets immediately)

  Note:

  Active calls between IP Phones are maintained for a limited time, but no call modification is available and, when the call is completed, the IP Phones are restarted.

  The backup system attempts to once again redirect the IP Phones to Site 1. When the redirection attempt fails, the IP Phones remain registered at Site 2, where they obtain normal service. While they remain registered at Site 2, the IP Phones display "Local Mode".

  When connectivity to Site 1 is restored, the IP Phones are automatically redirected again to their home system.

  Note:

  Established calls are completed before the redirection to the home system.

- Site 1 = Backup system: If Site 1 is their backup system (S1), the IP Phones registered at Site 2 continue to receive normal telephone service. If an IP Phone is reset, it attempts

to connect to Site 1 a number of times, defined by the S1 Retry Count of the IP Phone. When the S1 Retry Count of the IP Phone has expired, the IP Phone uses its S2 value to register directly to Site 2.

Note:

Unlike the 1+1 configuration, the Controlled Load-sharing configuration does not offer a transition of states on the backup system.

When the IP Phones are registered on Site 2, users have full access to the features and services configured on that system.

Site 2 system failure

Site 2 system failure operates in a similar manner to Site 1 system failure, with the exception that Site 1 assumes control of the IP Phones.

Figure 38: Load-sharing configuration: system failure at Site 1 on page 119 shows a Controlled Load-sharing configuration that is experiencing a failure at Site 2.

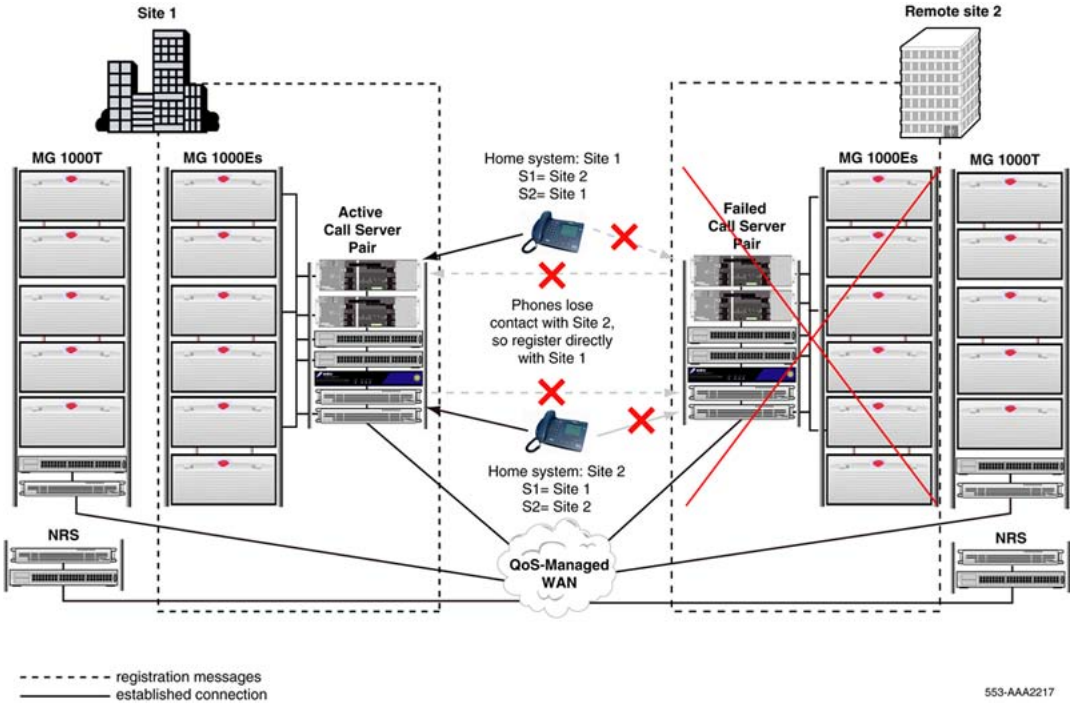

Figure 39: Controlled Load-sharing configuration: system failure at Site 2

Planning a Controlled Load-sharing configuration

To implement the Controlled Load-sharing configuration, plan each system to service all of the home and backup telephones that are to be defined on that system. That is, plan the system as though all telephones, including local telephones and redundant IP Phones that are going to be redirected to the opposite site, are registered at the local site. This ensures that the required capacity and services are available to provide redundancy in case the opposite site fails.

The home and backup systems must share the 3-digit NODE ID prefix, and redundant IP Phones defined with the same TNs and DNs on the home and backup systems.

Finally, each system can be planned independently to provide different features and services. If one system is configured with less hardware or fewer features and services, ensure that it is acceptable for the affected IP Phones to have less redundant capacity.

Additional planning considerations

The Controlled Load-sharing configuration provides redundancy only for IP Phones. Therefore, only those circuit cards that support the operation of IP Phones can be configured for redundancy. This includes TDM equipment for providing digital media services to the IP Phones, such as music and Integrated Recorded Announcer.

Each system can be configured to provide local service to analog (500/2500-type) and digital telephones. However, these telephones are inoperable when the local system fails.

Signaling Servers

Both home and backup Avaya CS 1000E systems or Avaya CS 1000M Systems must have Signaling Servers installed to provide service to IP Phones. The Signaling Servers at each site must be configured independently.

Signaling Server requirements are site-specific and can vary in number and configuration at each site. If one site has lower traffic and capacity requirements, it can have fewer Signaling Servers installed, provided they can handle the traffic created when the other system fails.

For more details on Signaling Server capacity, see *Avaya Communication Server 1000E Planning and Engineering, NN43041-220* and *Avaya Signaling Server Installation and Commissioning, NN43001-125*.

Enhanced Redundancy for IP Line nodes

To allow IP Phones to register to the different nodes at the home and backup sites, the Controlled Load-sharing configuration uses the Enhanced Redundancy for IP Line nodes feature. This feature relaxes the checking performed by a node on the Node ID that is presented by a registering IP Phone. It allows an IP Phone with a three-digit Node ID to register to a node that is configured with a four-digit Node ID. To enable the registration, the three-digit Node ID of the IP Phone must match the first three digits of the node's four-digit Node ID. For more information, see *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

For example, if the home system node number is 1110, the backup system must use the same leading three digits, 111, in its node number (for example, 1115). The IP Phones must be configured with the same three digits, 111, when defining its Node information. The IP Phone can connect successfully to both Node ID 1110 and 1115. In fact, it can connect successfully to any Signaling Server that has 111 as the prefix of its node number.

NRS

To support the IP Phone redirection process and to ensure that each system can properly route calls following a system failure, a Primary and Alternate Network Redirect Server (NRS) are required on the network. The Alternate NRS periodically synchronizes its database with the Primary NRS. This ensures that, if the Primary NRS fails, the Alternate NRS can assume the role of the Primary NRS.

To provide the necessary redundancy, each NRS must be installed in a different location. This can be accomplished by installing one NRS with each system or by installing the NRS in different remote locations. Wherever they are installed, you must ensure that at least one NRS remains operational following failure of either system.

For information on the required NRS routing entries for the Controlled Load-sharing configuration, refer to Numbering plan on page 125. For additional information on installing and configuring the Primary and Alternate NRS refer to *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

NCS

The Network Connect Server (NCS) is an application associated with the NRS that supports Geographic Redundancy. It allows the backup system node TPS to query the NRS directly to perform the redirection of IP Phones to the home system.

To support the Controlled Load-sharing configuration, the NCS properties must be configured when the home and backup endpoints are defined on the NRS and when the home and backup system IP telephony nodes are defined in Element Manager. For more information, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313* .

Vacant Number Routing

Vacant Number Routing (VNR) must be configured on each backup system. If a DN is not valid, the number is considered vacant by the system call processor, and VNR is used to route the call to the NRS for resolution.

Network Time Protocol

The Network Time Protocol (NTP) is used to synchronize computer clocks in the global IP network. It provides a comprehensive mechanism to access the national time and frequency dissemination services, organize the time-synchronization subnet and adjust the local clock in each participating subnet peer.

Because the parameters for NTP are dependant upon the geographic location, the NTP parameters stored on the Primary Call Server cannot be replicated to the Secondary Call Servers located in a different geographic location. The NTP parameters are stored in a separate database file (ntp.db) which does not get replicated to the Secondary Call Servers. Therefore, an NTP configuration does not survive a geographic redundancy switchover.

For more information about Network Time Protocol, see *Avaya Features and Services Fundamentals, NN43001-106*.

Time of day clock

The date and time display on each IP Phone is determined by the system to which the IP Phone connects. In the case of system failure, the idle clock on the IP Phone display must be localized to the correct time for the geographic location of the IP Phone.

Because the systems in a Controlled Load-sharing configuration can be located in regions with different time zones, the configuration supports the Branch Office feature that enables a different time zone to be specified for telephones at multiple sites. Each site must have the appropriate time zone configured for the IP Phones at the other site. In the case of system failure, the time zone adjusts the system time for display at the appropriate site. Idle IP Phones display the correct time for their area.

For more information, see Controlled Load-sharing zones on page 183.

User licenses

In the Controlled Load-sharing configuration, User licenses are required for all IP Phones that a system is backing up in addition to the telephones that are registered locally on the system.

The following user licenses are available:

Temporary IP User license

This IP licensing enhancement limits the number of IP Phone Terminal Numbers (TN) with Network User ID (NUID) configured. These TNs are not included in the total number of license limitations for IP User, Basic IP User, or ACD agent usage, which makes this option more cost-effective than Basic IP User license or IP User license.

Basic IP User license

This license counts IP Phones type 2001 (LD 11, TYPE 2001P2).

IP Users license

This license counts IP Phones type 2002, i2210/2211, 2004 and IP Softphone 2050 (LD 11, TYPE i2002, i2210, i2211, i2004, or i2050). If insufficient Basic IP User licenses are available for the IP Phone 2001, the IP User license can also be used for configuration of the IP Phone 2001. When IP User licenses are used to configure the IP Phone 2001, an error message is generated, recommending the purchase of additional Basic IP User licenses.

More information on user licenses for IP Phones is found in *Avaya Signaling Server IP Line Applications Fundamentals, NN43001-125*.

Data network planning

The home and backup systems must comply with network requirements as described in *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*. The home and backup systems use a common NRS for routing.

NUID

The Network user ID (NUID) for each IP Phone must match the IP Phone's dialable DN at the home system.

If the NUID is not configured, the IP Phone registers directly on the backup system. If the IP Phone is configured with a NUID and Network Home TN, the IP Phone is automatically redirected to the home system TPS and then to the home office system.

An NUID has a maximum of 15 digits. Under the Uniform Dialing Plan (UDP), the NUID consists of the Access Code (AC1/AC2 of backup system), the Home Location (HLOC of home system), the Location Code (LOC) and the home system DN, (for example, 6 343-5555). Under the Coordinated Dialing Plan (CDP), it can be an extension (for example, 4567).

Note:

The home system DN must be an Electronic Switched Network (ESN)-compliant DN.

To create the NUID, use one of the following DN keys defined at the IP Phone's home system TN: ACD, MCN, MCR, PVN, PVR, SCN or SCR.

Comments? infodev@avaya.com

Note:

If ACD (Automatic Call Distribution) key is used to create the NUID, the ACD DN or Message Center DN must be used.

For more information about CDP and UDP dialing plans, refer to *Avaya Dialing Plans Reference, NN43001-283*.

Network Bandwidth Management

The Network Bandwidth Management feature allows bandwidth zones to be configured on a network basis. This enables codec selection and bandwidth allocation software to identify whether IP Phones or Media Gateways are physically collocated (in the same bandwidth zone) even though they are controlled by different Call Servers.

Numbering plan

The Controlled Load-sharing configuration is designed to work only if the two systems use a common dialing plan. Any other configuration is not guaranteed to work properly.

As well, the DNs and TNs configured on both systems for the redundant IP Phones must match.

The NRS must be configured to properly route IP Phone registrations to the home system. It must also be configured to route incoming UDP, CDP, and VNR calls to the appropriate active system.

UDP calls and IP Phone registration redirection

In the Controlled Load-sharing configuration, the HLOC value for the home system endpoint must be configured on the NRS with the least-cost factor (that is, 1). This ensures that incoming UDP calls are directed to the home system while it remains operational.

In addition, the separate HLOC value for the backup system endpoint must be defined on the NRS, also with the least-cost factor. This allows the NRS to direct incoming UDP calls to the backup system when the home system fails.

Note:

When the home system fails, the UDP dial-in numbers for redundant IP Phones are different (as a result of the different HLOC values of the backup and home systems).

Note:

The NRS uses a polling mechanism to monitor the system status of both the home and backup systems.

NUID redirections with UDP

If the NUIDs are based on the UDP dialing plan (that is, NUID = AC + HLOC + DN), the HLOC routing entry on the NRS used to route incoming calls to the home system also supports the registration redirection requests to the home system. When the backup system TPS queries the NRS with the NUID to determine the home system of the IP Phone, the NRS responds with the least-cost route: the home system.

CDP and VNR calls

To ensure that incoming CDP and VNR calls are routed appropriately, the matching range of redundant IP Phone DNs used on both systems must be defined appropriately on the NRS for the two endpoints.

To ensure that incoming calls are directed appropriately to the home system when it is active, the DN range must be configured on the home system endpoint with the least-cost factor (that is, 1).

To ensure that the NRS directs the calls appropriately to the backup system following home system failure, the same DN range must be defined on the backup endpoint with a higher cost factor (for example, 2).

NUID redirections with CDP

If the NUIDs are based on the CDP dialing plan (that is, NUID = DN), the CDP routing entry on the NRS used to route incoming calls also supports registration redirection requests to the home system. When the backup system queries the NRS with the NUID to determine the home system of the IP Phone, the NRS responds with the least-cost route: the home system.

Branch Office support

The Controlled Load-sharing configuration supports the Branch Office feature only when CDP BUID form is used. Then two levels of redundancy can be received and appropriate NRS definitions are:

- for home system home-corresponding CDP entry is defined with cost factor 1
- for backup system - cost factor 2
- for branch office - cost factor 3

Otherwise, if the BUID UDP form is chosen, then one level of redundancy can be received; only one of the systems involved in the load-sharing scheme is the home for the IP Phone. NUID cannot be at the home system for the IP Phone. This prevents additional redirections; that is, for the IP Phone redirected from the branch office, load sharing is not applied. If the

home system fails, the IP Phone reregisters at the branch office. All this is due to the fact that backup, home, and branch office systems have different home location numbers.

For Controlled Load-sharing configuration to support the Branch Office feature, the home and backup systems must both be configured as the home office for the MG 1000B, according to the instructions in *Avaya Branch Office Installation and Commissioning, NN43001-314*. For the MG 1000Bs to be redundant, each site must use the same TN and DN ranges for the MG 1000B IP Phones.

Figure 40: Controlled Load-sharing configuration: Branch Office support on page 127 shows an MG 1000B installed in the Controlled Load-sharing configuration.

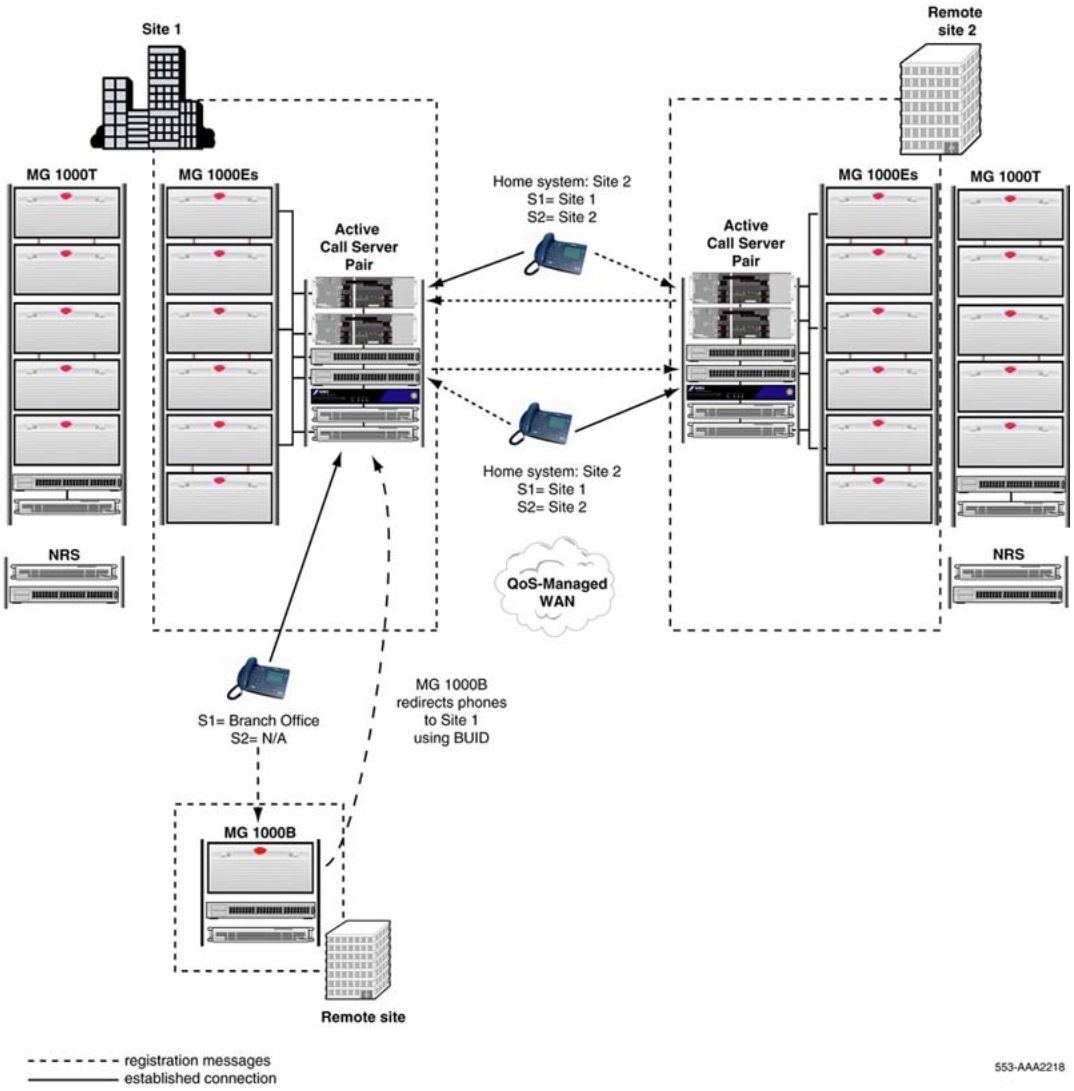

Figure 40: Controlled Load-sharing configuration: Branch Office support

The MG 1000B can be configured as survivable in conjunction with the Controlled Load-sharing configuration. In the unlikely event that both home and backup systems fail, the MG 1000B reverts to survivable.

NRS Routing for Branch Office

As the MG 1000B can operate with either the home or backup system functioning as the main office, the NRS must be configured to redirect MG 1000B IP Phones to the appropriate active system.

Note:

To redirect IP Phones to the main office, the MG 1000B uses the IP Phone's Branch User ID (BUID) value (configured in LD 11).

The NRS must also be configured to properly route incoming UDP, CDP, and VNR calls when the MG 1000B IP Phones are registered to the home system, to the backup system, or to the Branch Office.

UDP

When the MG 1000B IP Phones are registered on the home or backup system, the NRS definitions required to properly route incoming UDP calls are the same as those described in UDP calls and IP Phone registration redirection on page 125. Specifically, on the NRS, the HLOC for the home system and backup system endpoints must be defined with the least-cost factor.

In the unlikely event that both the home and backup systems fail, an additional NRS definition is required to ensure survivability of the MG 1000B. The branch office HLOC must be defined on the branch office endpoint with the least-cost factor. This additional entry ensures that UDP calls can reach the MG 1000B when the home and backup systems are both unavailable.

Note:

Because the HLOC value at the branch office is different than the HLOC value of the home and backup systems, the UDP dial-in number to the MG 1000B IP Phones is different when both main office systems fail.

CDP and VNR

When the MG 1000B IP Phones are registered on the home or backup system, the NRS definitions required to route incoming CDP calls to the IP Phones are similar to those described in CDP and VNR calls on page 126. Specifically, the range of main office DNs used for the Branch Office IP Phones must be configured on the NRS as least-cost for the home system and higher-cost for the backup system. This ensures that incoming CDP calls to the MG 1000B IP Phones are routed to the active home system.

These CDP entries also support the VNR call rerouting coming from the MG 1000B.

In the unlikely event that both the home and backup systems fail, a third NRS definition is required to ensure survivability of the MG 1000B. The same DN range must be defined on the branch office endpoint with a higher cost factor than the home and backup system endpoints. This entry ensures that CDP calls are rerouted to the MG 1000B when the home and backup systems are both unavailable.

BUID redirections with CDP

To support the Branch Office feature in the Controlled Load-sharing configuration, the BUIDs must be based on the CDP dialing plan, (that is, BUID = DN).

Note:

As the home and backup systems have different HLOC values, the Controlled Load-sharing configuration does not support UDP-based BUIDs.

As a result, the routing entry on the NRS used to route incoming CDP calls also supports registration redirection requests for the MG 1000B. When the MG 1000B queries the NRS with the BUID to determine the home system of the IP Phone, the NRS responds with the active home system: home or backup.

Temporary IP User license

This IP licensing enhancement limits the number of IP Phone Terminal Numbers (TN) with Network User ID (NUID) configured. These TNs are not included in the total number of license limitations for IP User, Basic IP User, or ACD agent usage, which makes this option more cost-effective than IP User license or Basic IP User license.

If the NUID is removed from an existing TN, the Temporary IP Users total is decremented by 1, and the totals for TNs used, IP or Basic IP Users, and ACD agents (if applicable) increment by 1.

Conversely, if the NUID is added to an existing IP Phone TN, the total of Temporary IP User usage increments by 1, and the totals for TN usage, IP User or Basic IP user usage, and ACD Agent usage (if applicable) are decremented by 1.

IP Phones registered with TNs configured with NUID and NHTN are redirected to the Network Home. If the Network home is unreachable, the IP Phones stay in Local Mode in their Secondary Call Server. In this situation the IP Phones are subject to network wide redundancy Local Mode Licensing restrictions and treatments.

Installing a Controlled Load-sharing configuration

Installing Controlled Load-sharing configuration on page 130 describes the high-level steps required to install a Controlled Load-sharing configuration.

Installing Controlled Load-sharing configuration

1. Install or upgrade each system to the appropriate software.
2. Configure the two systems for IP Peer Networking, with a Primary NRS and Alternate NRS. For more information about IP Peer Networking, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313* .
3. Configure your IP Phone DNs and TNs at both sites.
4. At each backup site, configure the appropriate NUID and NHTN values for the redundant IP Phones, as well as Vacant Number Routing.
5. For each redundant IP Phone, configure the appropriate Connect Server values (S1 and S2).

Provisioning the IP Deskphones

For information about installing and configuring IP Deskphones, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310* and *Avaya IP Deskphones Fundamentals, NN43001–368*. Ensure the S1 of the redundant IP Deskphone points to the backup system node TPS and S2 points to the home system node TPS.

For information about configuring IP Deskphones using Dynamic Host Configuration Protocol (DHCP), see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

The NUID and NHTN must also point to their home system. Configure the NUID and NHTN values on the backup system in LD 11 as follows:

Table 20: LD 11: Configure NUID and NHTN

Prompt	Response	Description
REQ	NEW/CHG	Add new data or change existing data
TYPE	2001P2/2002P1/2002P2 2004P1/2004P2/2050PC /2007 1120/1140/1150/1165 1210/1220/1230	IP Phone type.
CUST		Customer number.
	0-99	Range for Avaya CS 1000M System and Avaya CS 1000E.
NUID		Network User ID: Dialable home system DN.
	aaaa	Network User Id. Enter X to delete.
NHTN		Network Home TN: home system TN.

Comments? infodev@avaya.com

Prompt	Response	Description
	l s c u	Format for Avaya CS 1000M System and Avaya CS 1000E where l = loop, s = shelf, c = card, u = unit.

Maintenance

The home and backup systems conform to the methods and procedures used for local and remote access, as defined for Avaya CS 1000M Systems and Avaya CS 1000E systems.

The Controlled Load-Sharing implementation preserves all existing VxWorks or Overlay support tools available on the Avaya CS 1000M System or Avaya CS 1000E solutions, except where otherwise noted in this document.

Configuring the Database

During normal operation, each system is the master of its own customer database. Perform all database modifications on the Primary system only.

Note:

Both customer databases in the Controlled Load-Sharing configuration are configured and updated manually. There is no database-replication process involved.

If all the offered features and services are required to be redundant, the backup system must be configured appropriately, and each change that is made on one system must be reflected on the other system.

IP Phone Test Local Mode

The IP Phone Test Local Mode can be used with the Controlled Load-Sharing configuration to register an IP Phone to its backup system. For more details, refer to IP Phone Test Local Mode.

Feature interactions

The redundant features available to an IP Phone are limited to the features that are offered on its backup system.

System monitoring

Avaya recommends implementing SNMP alarm management to monitor the health of home and backup systems. SNMP can be used to trigger alarms, for example, when any IP Phone de-registers from a system. For more information, see *Avaya Communication Server 1000 Fault Management—SNMP, NN43001-719*.

Chapter 7: Campus Redundancy

Contents

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Normal Operations on page 136

Network topology on page 140

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Downgrading a redundant system on page 153

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Description

The Avaya Communication Server 1000E (Avaya CS 1000E) system is a highly-scalable and robust IP PBX that offers support of IP-based applications using industry-standard SIP and H.323 interfaces, while providing an industry-leading set of telephony features and applications.

The Campus Redundancy feature provides the ability to separate the Avaya CS 1000E Call Servers in a campus environment for "campus mirroring". This feature enables two Call Servers, one active and one redundant, to be connected through an Ethernet network interface. The Avaya CS 1000E system allows campus mirroring to operate using other vendors' switching products in addition to the BayStack 470. See Third-party vendor switching equipment on page 144.

To separate the redundant Call Servers, the ELAN subnet and the subnet of the High Speed Pipe (HSP) may be extended between the two processors using networking equipment that provides layer 2 end to end connectivity.

Note:

Campus Redundancy is not supported for Avaya CS 1000M Systems.

The HSP is the interface used to provide:

- synchronization of the redundant Call Server's disk and memory subsystems
- sharing of "Health" information
- memory shadowing between the two Call Servers during graceful switchover operations.

The HSP uses one of the network interfaces on each of the CPUs. The HSP port is labelled "LAN 2" on the faceplate of the processor.

Note:

Contact the Avaya representative for information on the latest supported products.

If the two Call Servers are collocated, the HSP can be connected using a standard CAT5e or CAT6 crossover cable, limited to 100 meters in length.

High Availability (HA) package

Campus Redundancy includes the addition of the 410 HIGH_AVAIL HIGH AVAILABILITY package for CP PM servers. If the HA package is present in the keycode, the CP PM Call Server uses the HSP to detect the presence of the other core. If the other core is detected, both cores negotiate to determine which is the active core and which is the standby core. If the CP PM Call Server cannot detect the other core, it comes up as a single core system.

If the HA package is not present in the keycode, the existing Call Server software is modified to block the HSP connection so that the CP PM Call Server does not attempt to detect the presence of the core. In the absence of the HA package, the CP PM Call Server runs as a single core system, even in a system with two cores and the HSP ports on both cores are connected.

Note:

In CS 1000 Release 6.0, the CP PM Co-res CS and SS does not support an HA configuration. For systems that require HA configuration, the VxWorks-based Call Server software must be deployed.

High Speed Pipe (HSP) IP address management

This allows IP addresses to be configured for the HSP that do not conflict with other addresses in the customer's enterprise IP networks.

These addresses are provisioned against a specific CPU (side 0 or side 1). Optionally, you may choose not to configure the HSP IP addresses and in this case the HSP interfaces use the following default IP addresses:

1. 127.2.0.1 is bound to the Side 0 CPU.
2. 127.2.0.2 is bound to the Side 1 CPU.

When configuring a HSP in which the CPUs are connected through a data network, the IP addresses must be configured for each CPU/Side so that address conflicts can be avoided within the IP network.

Stop and Copy protocol

The Stop and Copy protocol is used during a graceful switchover between CPUs. The switchover can be due to a manual request, midnight maintenance routine, or a health count mismatch between the two CPUs during regular traffic periods.

Stop and Copy includes a checksum mechanism to ensure mirror imaging of data on both cores of a redundant system. The checksum mechanism does the following:

- ensures data validity in both cores
- protects data from corruption
- reduces the time and process of Stop and Copy, thereby reducing graceful switchover time by approximately 25%
- reduces graceful switchover failures

If the checksum does not match, protected data is copied as part of Stop and Copy.

The system is restored to fully redundant mode after a failed Stop and Copy. When redundancy cannot be realized, the cores are split and the previously active core continues to be operational.

Database replication and restoration

For information about database replication and restoration, see

Operating parameters

The use of VLAN configurations and port priority settings to protect the ELAN and HSP network interfaces from harsh network conditions is required to ensure reliable operation. This minimizes the risk of unexpected network problems, such as heavy traffic conditions, broadcast storms, network stress caused by a virus, and Denial of Service attacks.

Fast Spanning Tree Protocol Learning (or disabling of Spanning Tree altogether), physical port-based priority, and VLANs must be supported by the networking products used to carry the HSP traffic. Avaya recommends Multi-link Trunking to provide redundancy in the connections between Avaya CS 1000E CPU locations.

Refer to HSP configuration on page 146 for details on the operating parameters.

Normal Operations

During normal operation the active and redundant CPUs are both running in parallel. The difference between the two CPUs is that the active side is running call processing applications while the redundant side is running only diagnostics and management tasks.

The redundant CPU can take over from the active CPU in certain conditions. The Campus Redundancy feature allows for a second "redundant" Call Server to take over from the "active" Call Server in certain conditions. System operation under Campus Redundancy is similar to the existing CP PII and CP PIV redundant call processor operations. During normal functioning, one Call Server is in active mode and the other is in redundant mode. The active Call Server's protected memory and disk subsystems are shadowed to the redundant Call Server using a synchronization protocol over a the 100BaseT HSP interface. This ensures that the redundant Call Server can assume system control in case of failure of the active Call Server.

Depending upon the situation a graceful switchover or an ungraceful switchover may occur. The active CPU becomes redundant and the redundant CPU becomes active.

Disk shadowing also allows for a graceful switchover, where system control can be switched from the active to the redundant CPU Call Server, while maintaining all call state information with only a brief pause (several seconds). The switchover can be performed once a day, during the midnight routines. This ensures that both Call Servers are fully capable of supporting call processing operations. Any failure in this switchover is reported with an error message and SNMP alarm, so that the problem can be corrected before becoming a hidden second point of failure.

Graceful switchover can also be performed manually to facilitate maintenance operations and software upgrades.

The two Call Servers also use the HSP to communicate their individual status (active or redundant) to each other. In addition, a health count is maintained within each Call Server for critical components such as disk drives, physical 100BaseT LAN ports, and connectivity to the other devices on the ELAN subnet. The health count is communicated between the two Call Servers and is used to detect whether one processor is healthier than the other.

The following sections describe the operation in more detail.

Warm start and Cold start

Warm start

This recovery mechanism involves restarting the entire Operating System on the Control Processor. All tasks are restarted. The call processing task (tSL1) executes the INITIALIZE procedure. Basically, all its protected configuration data is preserved (i.e. not reloaded from hard drive) and the task rebuilds all the unprotected data. In addition, established calls are rebuilt in the INITIALIZE procedure.

Cold start

This recovery mechanism involves restarting the entire Operating System. All tasks are restarted including the tSL1 task. The tSL1 task executes the SYSLOAD procedure. Basically, protected data is all cleared. Configuration data is reloaded from the hard drive and rebuilt into the protected data.

Fault Detection

Software watchdog mechanism

Important tasks are registered with the software watchdog task. The registered task has to "punch" the software watchdog periodically; otherwise, the corresponding software watchdog times out. In general, if the watchdog timer for a registered task times out twice consecutively, the registered task is restarted. If the task is restarted 15 times, a system warm start is invoked.

Hardware watchdog mechanism

A hardware watchdog is the defensive mechanism against processor runaway or a high priority task consuming all CPU cycles. If the hardware watchdog times out the first time, it triggers a system warm start. If the warm start process cannot reset the hardware watchdog, the second hardware watchdog triggers a system cold start.

Exception handling

When a processor exception occurs, the exception handler analyzes the cause of the exception and the context in which the exception is invoked. If a task causes an exception, the task is restarted. If the exception is from tSL1, system warm start is invoked.

Health Monitoring

Each Call Server possesses a health count (available via the STAT HEALTH or STAT CPU command in overlay 135). The Health State of each Call Processor Unit (CPU) is an indication of the current operating state of all the major hardware components in the network. Each component is assigned a health count based on the current operating state. The overall CP health measure is the sum of the health counts of all the components. As such, the CP health measure depends on the system configuration.

Switchover

Graceful Switchover

In normal operation the health count of each CPU should be equivalent. In the case where the active CPU detects that the redundant CPU has better health, a graceful switchover is invoked. In this process, almost the entire memory image from the active CPU is copied over to the memory of the redundant CPU. The redundant CPU resumes the operations left off from the active CPU after going through a post-switchover procedure. This post-switchover procedure includes sending out a gratuitous ARP message to the IP world for informing where the active IP ELAN address is located. This CPU becomes the active side.

The previously active side invokes a warm start after the copying operation is completed. After the warm start, it becomes the redundant side.

During a graceful switchover, there is usually no impact to calls already in progress. There is a brief duration whereby new calls are not allowed in the neighborhood of 6-8 seconds depending upon the configuration.

Graceful switchover may be invoked manually using the SCPU command in overlay 135.

Ungraceful Switchover

When it is decided that the active side is inoperable (e.g. power or processor failure, watchdog timeout, exceptions), the redundant side warm starts and takes over control. The switchover does not occur immediately, because when the redundant side detects loss of heartbeat, it

must wait long enough to be sure that the active side is not simply performing a warm start (INI). The timer used to invoke the ungraceful switchover is in the order of 56 seconds.

Midnight Switchovers

Because of the robustness of the new call processor architecture, the need to perform CPU switchovers during midnight routines are no longer required and have been removed from the software routine. With Call Processor PII and Call Processor PIV, both CPUs are alive (although only one is in call processing state).

It is still possible to have a daily switchover of the CPUs during the midnight routine if so desired. The following steps are required to provision this.

Important:

- If the customer can tolerate a service interruption of 6 to 10 seconds during the daily diagnostic routines which are normally scheduled outside of business hours (nominally “midnight”), and if the system is monitored daily or in real time for critical alarms, then the daily scheduled switchover of active CP side is recommended.
- If the customer cannot tolerate a daily scheduled diagnostic test, then a CP side switchover test should be conducted during a scheduled maintenance window at least once a year, and preferable monthly or quarterly.
- Never switching active CP side is not recommended, because a hardware fault on the inactive standby CP side may not be discovered until the active side experiences a hardware failure, a power failure, or a network connection failure.

  These recommendations apply to all dual processor machines.

- LD 17
- REQ chg
- TYPE ovly
- DROL 135
- MID_SCPU yes

Impact on calls

Depending upon whether the switchover is graceful or ungraceful there may be an impact. To for more information about the impact of switchover on calls, see *Avaya Converging the Data Network with VoIP Fundamentals, NN43001-260*.

Generally, established basic calls survive both the graceful and ungraceful switchovers. Basic calls that are in a transient state (for example, calls that are in the dialing state) survive a graceful switchover but do not survive an ungraceful switchover.

Conference call survivability

Conference calls on the Avaya CS 1000E system are rebuilt after system initialization, including initialization due to an ungraceful switchover of cores. This restores all established calls involved in add-on conference on the Avaya CS 1000E using network control memory. Any periodic tones in the conference call are dropped. The operation of Meet-Me conference calls is not affected. If a call requires allocation of a Physical Terminal Number (PTN) and is unable to get one after Call Server restart, the call is dropped.

ACD Queue Call Recovery

ACD Queue Call Recovery (ACDR) restores transient calls in ACD queues during CPP ungraceful switchover. Data for transient calls is mirrored on the inactive core and used to rebuild ACD queues after a system initialization resulting from an ungraceful switchover. A maximum of 1000 calls can be restored. If another system initialization is invoked (INI within INI) during system initialization, ACDR continues to restore calls. However, if another system initialization is invoked (INI within re-INI), ACDR aborts and an INI0112 message prints on the maintenance TTY.

During a Call Server warm start, the MSDL gets disabled and re-enabled and the MSDL300 message prints on the maintenance TTY. The D-channels are also disabled and re-enabled, dropping the transient calls. If an ACD call is not recovered before the MSDL300 message prints, queued calls with ringback treatment are dropped. If the queued calls are provided with RAN/MUSIC treatment, the calls are not dropped and can be recovered as these calls are already established.

Heartbeat

The two CPUs exchange heartbeats to determine if the other CPU is reachable over the HSP. The heartbeat protocol also carries information regarding the health count of each CPU. If the HSP is disconnected then the heartbeat protocol attempts to traverse the ELAN instead

If the heartbeat cannot be communicated between the two CPUs meaning that connection over the HSP and ELAN is lost between the two CPUs then the redundant CPU warm starts to become active after a certain period of time.

By optimizing timeout and threshold parameters used in retries of the heartbeat mechanism, ungraceful switchover trigger time is reduced to less than 15 seconds. The optimization in the timing leads to a change in the INI policy. When the active core warm starts, the inactive core also reboots, so no swapping of the cores takes place.

Network topology

The Avaya CS 1000E system provides the ability to distribute the redundant Call Server CPUs to two locations.

The initial offering of this feature in previous releases made use of dark fiber driven directly by BayStack 470 Layer 2 switches. This allowed the Avaya CS 1000E redundant Call Servers to

be distributed to two locations that are separated by as much as 40 km. This configuration, see Figure 41: Call Server and Signaling Server (HSP and ELAN subnet) separated with Layer 2 switching products on page 142, is still supported as the base offering. The Campus Redundancy enhancements starting in Release 4.5 supports any vendor's switching product, providing an installation test is run to measure packet loss, jitter, and delay.

Baystack 470 GBIC Fibre Interfaces

The Campus Redundancy feature is supported in the Release 4.0 timeframe using Baystack 470 Layer 2 switches only as the transport mechanism between the 2 system cores.

Any of the Baystack GBIC interfaces can be used as long as the interface specifications are met for cable type, length and attenuation. The following are different Baystack 470 GBIC's available.

- 1000BASE-SX on MultiMode Fiber (50 ?m) 550 m
- 1000BASE-SX on MultiMode Fiber (62.5 ?m) 275 m
- 1000BASE-LX on MultiMode Fiber (50 ?m) 550 m
- 1000BASE-LX on MultiMode Fiber (62.5 ?m) 550 m
- 1000BASE-LX on SingleMode (9 ?m) 5 km
- 1000BASE-XD on SingleMode (9 ?m) 40 km

Campus Redundancy Baystack 470 Bandwidth Use

The Avaya Communication Server 1000E ELAN and HSP Ethernet links each require a dedicated 100 Mbps VLAN on the Baystack 470 1Gbps link. Although not specifically stated in the documents, the remainder of the Baystack 470 1Gbps Fiber link (800 Mbps) can be used for other data.

The requirements to use this extra bandwidth include:

- The extra bandwidth must be configured on a VLAN separate from the ELAN and HSP VLANs
- The ELAN and HSP VLANs must be configured with higher priority than the other VLANs to ensure they get bandwidth when required for an HSP Call Server switch over.

Avaya recommends that the actual configured aggregate bandwidth for the extra data traffic not exceed 800 Mbps. This further ensures that the ELAN and HSP ports always have enough bandwidth to complete their tasks.

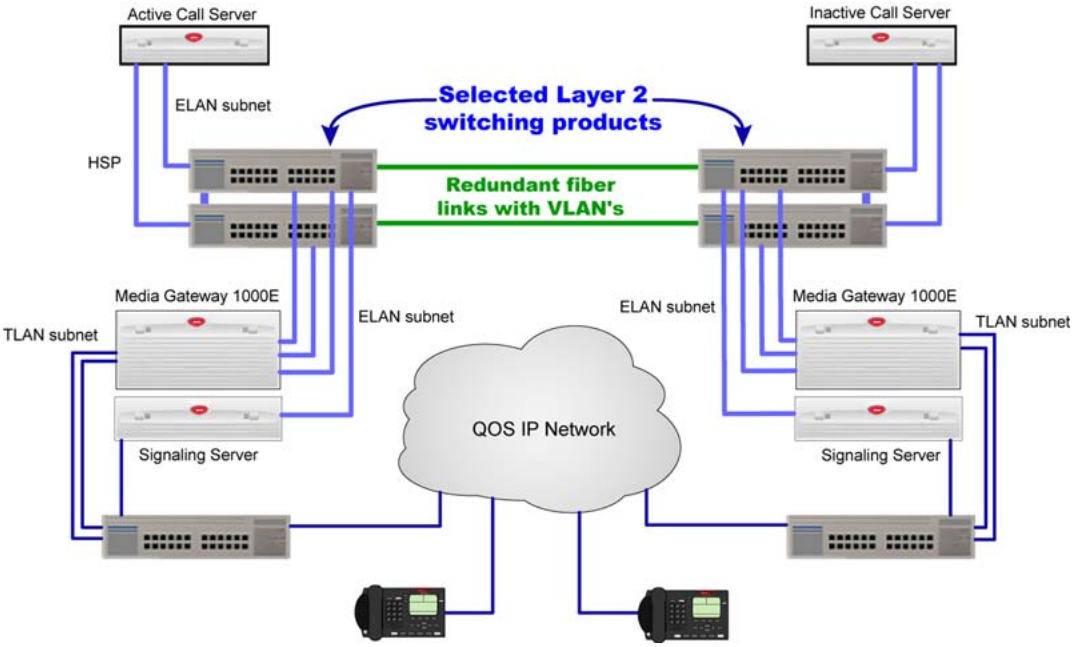

Figure 41: Call Server and Signaling Server (HSP and ELAN subnet) separated with Layer 2 switching products

In addition, various other network topologies are supported that allow converged voice and data traffic to share the links between locations. However, the network design must include mechanisms such as priority, isolated VLANs, and other robust design principals to ensure that the network design meets stringent requirements for packet loss, delay, and other network characteristics.

Figure 42: Call Server and Signaling Server (ELAN subnet and HSP) separated by high capacity network on page 143 illustrates an alternative network topology that supports the Campus Mirroring of Avaya CS 1000E CPUs.

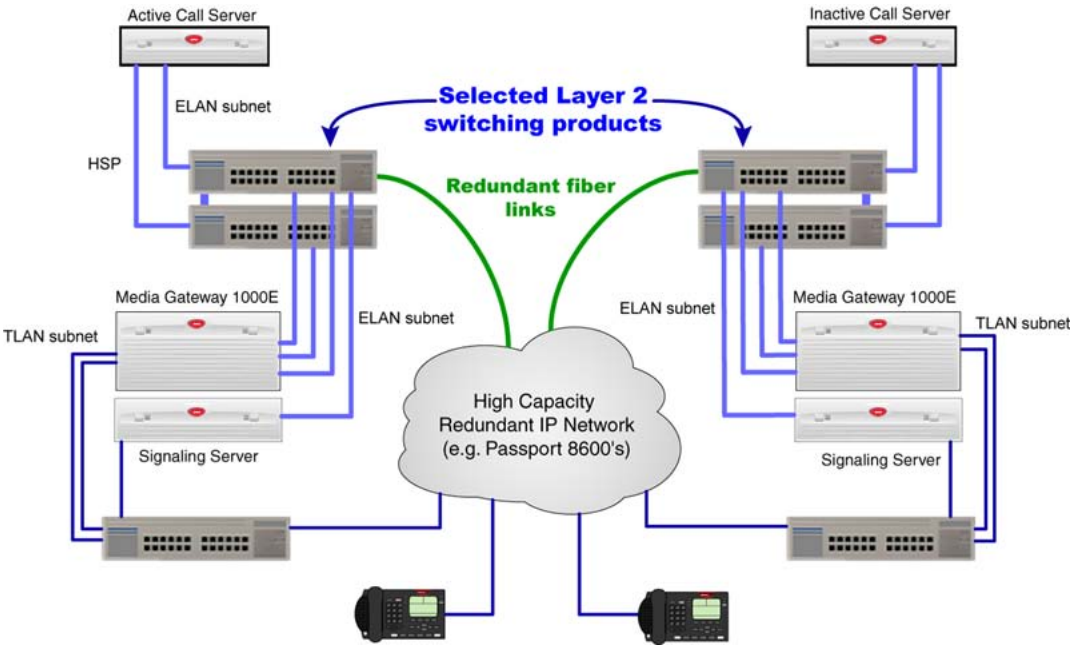

Figure 42: Call Server and Signaling Server (ELAN subnet and HSP) separated by high capacity network

Note:

Avaya recommends that Node Terminal Proxy Server (TPS) Signaling Servers be distributed between the two sites to prevent the system from disabling, in case of any local disaster in a single site (that might trigger failure of TPS registration). Configure the Signaling Servers in both sites in a single node to enable load balancing between all Signaling Servers in the system.

Switching Equipment

Layer 2 switching equipment

The following equipment supports both the MLT (Multi Link Trunking), port based VLANs, and 802.1P priority configuration and is recommended for the HSP application.

- 325-24T - Layer 2 VLANs, MLT, 802.3ad
- 325-24G - Layer 2 VLANs, MLT, 802.3ad
- 425-24T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 425-48T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 460-24T-PWR - Layer 2 VLANs, MLT, DMLT, , 802.3ad, 802.3af PoE

- 470-24T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 470-48T - Layer 2 VLANs, MLT, DMLT, 802.3ad
- 5510-24T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 5510-48T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 5520-24T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing, 802.3af PoE
- 5520-48T - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing, 802.3af PoE
- 8300 - Layer 2 VLANs, MLT, DMLT, L3 interVLAN routing
- 8600 - Layer 2 VLANs, MLT, DMLT, SMLT, 802.3ad, L3 interVLAN routing

Third-party vendor switching equipment

The HSP supports any vendor's switching equipment.

The following third party equipment has been tested:

- CISCO WS-3750G 24T-E GE ENH MULTILAYER CAYALYST (Layer 2 VLAN mode)
- 3C17203-3COM US/ 3COM 24-PORT 10/100TX SWITCH W/2
- 3COM 3C17304-US 3COM SS3 SWITCH 4228G 28PORTS EN
- 13240 EXTREME SUMMIT 200-24 SWITCH - 24 PORTS

Note:

The HSP cannot be routed. This means that the HSP cannot be extended through a layer 3 router unless that device supports a method of providing layer 2 end to end connectivity ie. layer 2 tunnelling. Therefore, when passing through routing equipment, the HSP must remain in the same subnet from one Call Server to the other (for example, tunneling the HSP over the network).

Call Server operation during IP network failure

The following section describes the operation of the Avaya CS 1000E system during various failure scenarios, and how the system recovers after the network is properly restored.

ELAN subnet connectivity between the CPUs is lost but HSP is still operational

In this scenario, only one CPU binds the Primary ELAN network interface IP address to its network interface. The other CPU binds the Secondary ELAN network interface IP address (no IP address conflict). All Media Gateway 1000Es are controlled by the CPU that has the

Primary ELAN network interface IP address. However, the Signaling Servers and Voice Gateway Media Cards are included in the health count of the CPUs, and each of these attempt a connection to both the Primary ELAN network interface IP address and the Secondary ELAN network interface IP address.

It is possible that, within minutes, the health counts of the CPUs can change, so that the CPU with the largest number of possible connections to ELAN devices has a higher health count than the other CPU. This causes a CPU switchover to occur, again with only one CPU binding the Primary ELAN network interface IP address.

HSP connectivity is lost but ELAN subnet connectivity between the CPUs is operational

In this scenario, the same health count is calculated based upon connectivity (and other health count factors). This health information is now shared over the ELAN subnet. The CPUs are still redundant but memory and disk space are not synchronized between the active and redundant CPU.

ELAN subnet and HSP connectivity is lost between the CPUs

In this scenario, the subnet is split into two broadcast domains/segments, each holding one CPU and a number of Media Gateway 1000Es. Specifically, in this scenario. Each CPU attempts to talk to the other CPU. When the HSP connection is restored then health information is exchanged. The CPU with the lower health count reboots to become redundant.

In this case, the CPUs and Media Gateway 1000Es split into two independent systems, (each CPU is active) providing services to all their registered IP Phones and voice gateways, independent of each other. When a network connection is restored (either HSP or ELAN subnet), CPU 1 reboots and allows CPU 0 to resume complete control of all Media Gateway 1000Es.

Note:

ARP is Address Resolution Protocol, the TCP protocol that translates an IP address into the MAC (physical hardware) address of the card. The ARP performs the address resolution.

HSP configuration

Initial installation

The HSP IP addresses are not configured during installation. Instead, they are configured in LD 117 after the system comes up. If HSP IP addresses have already been manually configured, then they are used as the system reboots. Otherwise, the default HSP addresses are used.

HSP recommendations and rules

The following are recommendations and rules for configuring the HSP network interface and network when using network equipment to connect the HSP network interfaces of the two Call Servers.

- The HSP must be connected through a cross-over cable or by a dedicated VLAN through switches.
- The HSP must be in its own IP subnet. It cannot be combined with the ELAN subnet.
- The minimum throughput of the HSP must be 100 MB. Therefore, the HSP port must be 100 MB and full duplex. This must be confirmed using the **STAT HSP** command in LD 137 after the equipment is operational. This must also be verified on the network equipment to which the HSP is attached.
- The network switches must be capable of port mapping to 802.1p/Q.
- When running the HSP across network equipment, the HSP must be isolated in its own VLAN. Do not include other traffic in this VLAN. This VLAN must be given higher VLAN priority than any other traffic on the network, except for network control traffic (network control traffic is the traffic necessary to keep the network operational). The VLAN must be 802.1p/Q-capable and be configured to a very high setting so as not to starve the HSP. Avaya strongly recommends 802.1p Level 7 (Network Control and OAM).
- When using third-party vendor network equipment that has not been validated by Avaya, a pre-test of the network must be performed. This test includes mixed traffic going across the networks in different VLANs. The network specifications should meet the round trip delay and packet loss requirements.
- The round trip delay of the HSP VLAN must be less than 30 msec and the packet loss of the HSP VLAN must be below .1 % packet loss. See Customer validation on page 151.
- The HSP port on the CPP4 is configured to auto-negotiate the link speed and duplex. Therefore, the network equipment to which the CPP4 is attached must also use auto-

negotiate. Verify that both the CPP4 and the network equipment speed and duplex are a match. The CPP2 does not auto-negotiate; instead, it is fixed to 100 Mbps and full-duplex. Verify that both the CPP2 and the network equipment speed and duplex is a match.

- Avaya recommends that MLT (Multi Link Trunking) be used across the enterprise IP network for the Campus Redundancy configuration.

Caution:

Duplex mismatches occur in the LAN environment when one side is configured to Auto Negotiate and the other is hard configured.

The Auto Negotiate side adapts only to the speed setting of the fixed side. For duplex operations, the Auto Negotiate side sets itself to half-duplex mode. If the forced side is full-duplex, a duplex mismatch occurs.

High Speed Pipe IP address configuration

The configuration of High Speed Pipe (HSP) IP addressing can optionally be performed after the installation process, if the default IP addresses are not appropriate for the customer network. Avaya strongly recommends allocation of a network IP address within the customer's address space, if the network is not dark fiber-driven by BayStack470 switches.

Existing configuration procedures are used to provision these IP addresses. Specifically, Host Names are used to identify the HSP ports IP addresses. The names used are:

- DEV_SIDE0_HSP - host name for the HSP on the Side 0 CPU
- DEV_SIDE1_HSP - host name for the HSP on the Side 1 CPU

The HSP IP addresses do not have to be manually configured. They can still negotiate to default IP addresses automatically.

CPU Side 0 = 127.2.0.1 CPU Side 1 = 127.2.0.2

Note that the addresses are specific to a particular side.

The host names are not configurable; however, their parameters are configurable in LD 117. Changes to host name parameters require the use of the **`out`** and **`new`** commands; the **`chg`** command is not allowed.

Note:

The HSP IP addresses and subnet mask are not activated until the SET HSP_IP command is used or the CPU reboots.

Note:

The CPU should not be rebooted after changing but before issuing the SET HSP_IP command. Doing so may cause the HSP addresses between the active CPU and the redundant CPU to become out of sync.

Example of HSP configuration

```
=> new host DEV_SIDE0_HSP 192.168.100.10

INET Data Added

=> new host DEV_SIDE1_HSP 192.168.100.11

Warning: HSP Subnet Mask not configured. Please enter HSP Subnet mask
using the CHG HSP_MASK command

INET Data Added

=> chg hsp_mask 255.255.255.248

INET Data Changed

=> prt host

Call Server

ID Hostname?????IP Address

?1 LOCAL_PPP_IF???137.135.192.4

?2 REMOTE_PPP_IF???100.1.1.1

?3 ACTIVE_CPU???47.11.226.10

?4 INACTIVE_CPU???47.11.226.11

?7 DEV_SIDE0_HSP???192.168.100.10

?8 DEV_SIDE1_HSP???192.168.100.11

=> prt hsp_mask

HSP SUBNET MASK: "255.255.255.248"

OK

=> set hsp_ip

Activating HSP Addresses. Please wait ...

System is Redundant. Rebooting Inactive side to activate new HSP IP
addresses ...

24/03/2005 01:03:31 SRPT0118 CM: Server connection lost.

SRPT118 CM: Server connection lost.

Side 0 HSP IP set to "192.168.100.10"

Side 1 HSP IP set to "192.168.100.11"
```

```
HSP subnet mask set to "255.255.255.248"

OK
```

HSP IP address commands

The **NEW HOST** command is used to configure the HSP IP addresses.

Syntax:

```
NEW HOST DEV_SIDE0_HSP <ip address>
```

or

```
NEW HOST DEV_SIDE1_HSP <ip address>
```

The **PRT HOST** and **OUT HOST** commands are used to display and remove the host entries for HSP ports.

HSP subnet mask commands

The HSP subnet mask must be configured in LD 117 if the HSP IP addresses are configured. Three commands are available:

- CHG HSP_MASK
- OUT HSP_MASK
- PRT HSP_MASK

Table 21: LD 117 HSP subnet mask commands on page 149 describes the LD 117 HSP subnet mask commands.

Table 21: LD 117 HSP subnet mask commands

Command	Description
CHG HSP_MASK <subnet mask>	Modifies the manually-configured subnet mask, if it exists; otherwise, the subnet mask to the Call Server is added
OUT HSP_MASK	Removes the configured HSP subnet mask from the Call Server and replaces it with the default HSP subnet mask
PRT HSP_MASK	Retrieves the manually-configured HSP mask from the Call Server, if the mask exists, and displays it on the screen; otherwise, the default HSP subnet mask (255.255.255.0) is displayed

HSP IP address activation

The SET HSP_IP command is introduced to LD 117 to activate the HSP IP addresses and subnet mask.

Table 22: LD 117 HSP IP address activation command

Command	Description
SET HSP_IP	Activates the HSP IP addresses and subnet mask

The SET HSP_IP command first causes sanity checks to be performed on the configured HSP IP addresses and subnet mask. If the IP addresses and subnet mask are configured correctly, a warm Restart message is sent to the redundant side, if the system is redundant. Then the local HSP network interface is configured with the HSP IP address and subnet mask from the manually-provisioned parameters. Because the system is redundant, the HSP IP address parameters are copied to the redundant side, so that when the redundant side boots up, the new IP addresses and subnet mask are used.

If the system is not redundant, only the local interface is configured with the HSP IP address and subnet mask from the manually-configured values.

If the SET HSP_IP command is executed and the HSP IP addresses and subnet mask are the same as the IP addresses and subnet mask already in use, then this command has no effect.

HSP status command

The **PRT HOST** and **PRT HSP_MASK** commands print the configured values of the HSP IP address and subnet mask. These values are not necessarily the HSP IP address and subnet mask in use on the Call Server. These values are not applied until the **SET HSP_IP** command is issued successfully.

To determine the current HSP IP address and subnet mask in use on the Call Server, the **STAT HSP** command is introduced in LD 137.

Table 23: LD 137 status command

Command	Description
STAT HSP	Displays the current HSP IP address and subnet mask in use on the Call Server

The following is an example of the output from a **STAT HSP** command.

```
.stat hsp

LCS HSP STATE is UP
```

```
HSP LINK CARRIER: OK

Auto Negotiation: Enabled

Auto Negotiation Completed: YES

Actual Line Speed: 100 Mbps

Actual Duplex Mode: Full Duplex

Ethernet (gei unit number 1):

Internet address: 192.2.3.11

Broadcast address: 192.2.3.255

Ethernet address: 00:c0:8b:07:bd:fd

Netmask: 0xffffff00; Subnetmask: 0xffffff00

415607 packets received; 680621 packets sent

0 input errors; 0 output errors

0 collisions
```

Customer validation

If the customer chooses to use network equipment between HSP ports, then the following must be done:

- Prior to installation, the network Service Level Agreement (SLA) for the HSP must meet minimum requirements.
- The network must meet the minimum requirements. See HSP recommendations and rules on page 146.
- Call processor graceful switchover must be tested after the CS 1000 installation.

IP Telephony node configuration

If there is a significant risk that the IP network could lose connectivity between the two Call Servers and their surrounding IP Telephony node devices, such as the Signaling Server and Voice Gateway Media Cards, then there is a possibility that such a split can cause two of those devices to attempt to bind to the same node IP address while the network is split. If the network segments are separated from each other, this should not cause a problem; but when the network is restored, it can take up to a minute for the node to elect a new node master and restore full operation. To avoid this situation, configure separate IP Telephony nodes at each physical location, when the Server settings (S1/S2) for IP Phones are both available for use.

If either S1 or S2 is used for Geographic Redundancy or Survivable Branch Office, then each of the IP Telephony nodes can be registered with the Network Connect Server in the NRS with different cost factors, so that node redundancy is provided for these IP Phones as well.

Note:

If the TLAN network interfaces provided by these devices are to be split into different subnets, they must be configured in separate nodes, or the network must provide a VLAN broadcast domain between the two locations. This is another reason to use separate nodes when using Campus Redundancy.

Upgrading a redundant system

The Release 6.0 algorithm is backwards-compatible with the Release 5.5 algorithm, so when one side of a redundant system is upgraded, the HSP pipe continues to function.

When the remaining Release 5.5 Call Server is upgraded to Release 6.0 software, the new HSP algorithm takes effect and the HSP addresses are bound as described in High Speed Pipe (HSP) IP address management on page 134.

Note:

When upgrading from Release 5.5 to Release 6.0 software, the HSP IP address must not be manually configured before upgrading both Call Servers to Release 6.0 software.

The following table shows an example of upgrading a redundant system from CS 1000 Release 5.5 software to Release 6.0 software.

Table 24: Example of upgrading system software

Command	CPU 0	CPU 1
	State = Active/ Redundant HSP IP = 127.2.0.2 S/W = 5.5	State = Inactive / Redundant HSP IP = 127.2.0.1 S/W = 6.0
Active side (CPU 0): SPLIT	State = Active / Split HSP IP = 127.2.0.2 S/W = 5.5	Cold Start to become split State = Inactive / Split HSP IP = 127.2.0.1 (w negotiation) S/W = 6.0
Inactive Side (CPU1): 1) upgrade to 6.0 S/W 2) reboot -1	No change	Cold Start State = Inactive / Split HSP IP = 127.2.0.1(w negotiation) S/W = 6.0

Command	CPU 0	CPU 1
Active side (CPU 0): CUTOVR	State = Inactive / Split HSP IP = 127.2.0.2 S/W = 5.5	Warm Start to become Active State = Active / Split HSP IP = 127.2.0.1 (w negotiation) S/W = 6.0
Inactive Side (CPU 0): 1) upgrade to 6.0 S/W 2) reboot -1	Cold Start State = Inactive / Split HSP IP = 127.2.0.2 (w negotiation) S/W = 6.0	No change
Active side (CPU 1): JOIN	Warm Start to become redundant State = Inactive / Redundant HSP IP = 127.2.0.1(w negotiation) S/W = 6.0	State = Active / Redundant HSP IP = 127.2.0.2 (IP address corrected when JOIN is issued) S/W = 6.0
Active Side (CPU 1): 1) LD 117 ? enter HSP addresses for both sides and optional subnet mask 2) LD 117 SET HSP_IP	Warm Start State = Inactive / Redundant HSP IP=DEV_SIDE0_HSP (inet.db) S/W = 6.0	State = Active / Redundant HSP IP=DEV_SIDE1_HSP (inet.db) S/W = 6.0

Downgrading a redundant system

It is possible to have a redundant system running CS 1000 Release software with custom HSP IP addresses. When downgrading the software, one side (Call Server) reverts back to the default IP address and cannot talk to the other side. To avoid this situation, perform the following steps in the given order before performing the downgrade.

```
=> prt host

Call Server

ID Hostname?????IP Address

?1 LOCAL_PPP_IF???137.135.192.4

?2 REMOTE_PPP_IF???100.1.1.1

?3 ACTIVE_CPU???47.11.226.10

?4 INACTIVE_CPU ???47.11.226.11

?7 DEV_SIDE0_HSP???192.168.100.10
```

```
?8 DEV_SIDE1_HSP???192.168.100.11
=> out host 7
=> out host 8
=> set HSP_IP
```

split and downgrade

HSP maintenance

STAT CPU

The STAT CPU command in Overlay 135 provides the status of both CPUs in a redundant configuration. This command is only issued on the active CPU and gives an indication of the redundant CPU on a best effort basis (that is, if the system is not redundant the active side may not be able to communicate with the redundant side to get its status).

Definition of stat cpu results:

The first row indicates the state of the redundant system. It can be one of the following

- TRUE REDUNDANT - This means that both CPUs are up and actively communicating with each other. Disk and memory shadowing are complete.
- SPLIT HSP DOWN - The redundant system is split (the split command has been issued). Both CPUs are communicating over the ELAN but disk and memory shadowing between the two CPUs is not synchronized.
- SPLIT HSP UP - The redundant system is up but the system has been manually split by issuing the split command in overlay 135.
- REDUNDANT HSP DOWN - The system is redundant but the HSP is down. Both CPUs are communicating over the ELAN but the disk and memory shadowing between the two CPUs is not synchronized.
- SYNCING - The system is not redundant but disk and memoring shadowing between the two CPUs is in progress.
- SINGLE - The system does not have a redundant CPU or the ELAN and HSP to the redundanct CPU has been disconnected.

DISK STATE. The second row indicates the intended state of the disk. It can be one of the following:

- DISK STATE = SPLIT - This indicates that the administrator has issued the SPLIT command in overlay 135.
- DISK STATE = REDUNDANT - This indicates that the administrator has issued the JOIN command in overlay 135.

HEALTH. The HEALTH indicates the relative health of each CPU. This number varies depending upon the system configuration. If either CPU experiences problems with hardware or connectivity to the network then the health count goes down.

Table 25: Overlay commands and results

Action	Result
JOIN command in overlay 135	TRUE REDUNDANT DISK STATE = REDUNDANT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 1, DRAM SIZE = 256 MBytes TRUE REDUNDANT DISK STATE = REDUNDANT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 0, DRAM SIZE = 256 MBytes
SPLIT command in overlay 135	SPLIT HSPDOWN DISK STATE = SPLIT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 1, DRAM SIZE = 256 MBytes SPLIT HSPDOWN DISK STATE = SPLIT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 0, DRAM SIZE = 256 MBytes
Disconnect the HSP	REDUNDANT HSPDOWN DISK STATE = REDUNDANT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 1, DRAM SIZE = 256 MBytes REDUNDANT HSPDOWN DISK STATE = REDUNDANT HEALTH = 0 VERSION = Mar 22 2005, 15:21:44 Side = 0, DRAM SIZE = 256 MBytes
SCPU command in overlay 135	SYNCING DISK STATE = REDUNDANT HEALTH = 18 VERSION = Mar 22 2005, 15:21:44 Side = 0, DRAM SIZE = 256 MBytes SYNCING DISK STATE = REDUNDANT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 1, DRAM SIZE = 256 MBytes
Disconnect HSP and ELAN	? REDUNDANT HSPDOWN DISK STATE = REDUNDANT HEALTH = 20 VERSION = Mar 22 2005, 15:21:44 Side = 0, DRAM SIZE = 256 MBytes REDUNDANT HSPDOWN DISK STATE = REDUNDANT HEALTH = 18 VERSION = Mar 22 2005, 15:21:44 Side = 1, DRAM SIZE = 256 MBytes

STAT HSP

Use LD 137's STAT HSP command to monitor the HSP network interface.

The following is an example of Stat HSP output for a CP P4 processor (available only on the active side).

```
.stat hsp

LCS HSP STATE is UP

HSP LINK CARRIER: OK

Auto Negotiation: Enabled

Auto Negotiation Completed: YES

Actual Line Speed: 100 Mbps

Actual Duplex Mode: Full Duplex

Ethernet (gei unit number 1):

Internet address: 192.2.3.11

Broadcast address: 192.2.3.255

Ethernet address: 00:c0:8b:07:bd:fd

Netmask: 0xffffff00; Subnetmask: 0xffffff00

415607 packets received; 680621 packets sent

0 input errors; 0 output errors

0 collisions
```

STAT ELNK

The following is an example of the **STAT ELNK** command.

```
.stat elnk

ELNK ENABLED

Auto Negotiation: Enabled

Auto Negotiation Completed: YES

Actual Line Speed: 100 Mbps

Actual Duplex Mode: Full Duplex
```

```
Ethernet (gei unit number 0):
Host: PRIMARY_ENET
Internet address: 47.11.226.10
Broadcast address: 47.11.226.31
Ethernet address: 00:c0:8b:07:a5:9e
Netmask: 0xff000000 ; Subnetmask: 0xffffffe0
15 packets received; 20 packets sent
0 input errors; 0 output errors
0 collisions
New info
```

Troubleshooting

If one of the following problems occurs:

- the Call Servers do not perform graceful switchover and/or come up as single CPUs
- Disk sync and mem sync take a long time (greater than 10 minutes)

Then check the following:

- If it is a CP PIV processor, check the duplex and speed. Duplex mismatch is quite possible especially during an upgrade to CP PIV from CP PII using Baystack equipment. Duplex mismatch allows HSP to function, but packet loss is great.
- If it is a CP PII processor, verify that the Speed and duplex of the LAN equipment connected to the HSP is hard-coded to 100 Mbps full duplex.
- If the HSP traverses a network of switches make sure that the HSP is on its own VLAN. Verify that the 802.1 priorities are configured properly.

Appendix A: Configuring the BayStack 470-24T for Campus Redundancy

Contents

This section contains information on the following topics:

Description on page 159

BayStack 470-24T configuration on page 160

Description

Table 26: BayStack 470-24T VLAN assignment on page 159, Table 27: VLAN port configuration on page 160, and Table 28: MultiLink Trunk configuration on page 160 describe the configuration of the BayStack 470-24T VLANs.

Table 26: BayStack 470-24T VLAN assignment

VLAN Number	VLAN Name	VLAN Type	Port Membership
1	Default	Port-based	Switch 1- ports 1-26 (all) Switch 2- ports 1-26 (all) Switch 3- ports 1-26 (all) Switch 4- ports 1-26 (all)
2	HSP	Port-based	Switch 1- port 1, port 25 (GBIC) Switch 2- port 25 (GBIC) Switch 3- port 1, port 25 (GBIC) Switch 4- port 25 (GBIC)
3	ELAN	Port-based	Switch 1- ports 2-24, port 25 (GBIC) Switch 2- ports 1-24, port 25 (GBIC) Switch 3- ports 2-24, port 25 (GBIC) Switch 4- ports 1-24, port 25 (GBIC)

In this configuration example, the following ports in the BayStack 470-24T switches are connected to Call Server ELAN network interfaces (in VLAN 3: ELAN):

- Switch 2 – port 24
- Switch 4 – port 24

Different network interfaces can be assigned to the BayStack 470-24T VLANs and connected to HSP and ELAN network interfaces on each Call Server. The configured VLAN network interfaces must match the physical connections.

Table 27: VLAN port configuration

Switch Number	Port Number	PVID (Port VLAN Identifier)	Tagged Member
Switch 1	1	2	No
	2-24	3	No
	25-26	1	Yes
Switch 2	1-24	3	No
	25-26	1	Yes
Switch 3	1	2	No
	2-24	3	No
	25-26	1	Yes
Switch 4	1-24	3	No
	25-26	1	Yes

Table 28: MultiLink Trunk configuration

Trunk	Trunk Members [Unit /Port]
1	[1 / 25] [2 / 25]
2	[3 / 25] [4 / 25]

Note:

The four remaining high-speed fiber uplinks (GBIC ports 26) are not used in this configuration. They can be optionally used for MLT by being added to the existing uplink fiber network interfaces (ports 25), or for other dedicated uplink connectivity to network core switches.

BayStack 470-24T configuration

Configuring the BayStack 470-24T using Web-based management on page 161 must be performed for the switch stacks at both Call Server sites.

Configuring the BayStack 470-24T using Web-based management

1. From the main menu of the BayStack 470-24T Web-based management interface, choose **Configuration > Port Management**.
2. Configure the HSP network interface on switch 1 (and 3) as shown in Figure 43: HSP port configuration on page 161.

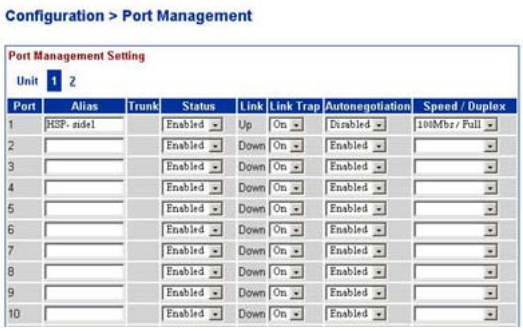

Figure 43: HSP port configuration

3. Configure the GBIC network interface on switch 1 (and 3) as shown in Figure 44: GBIC port (1) configuration on page 161.

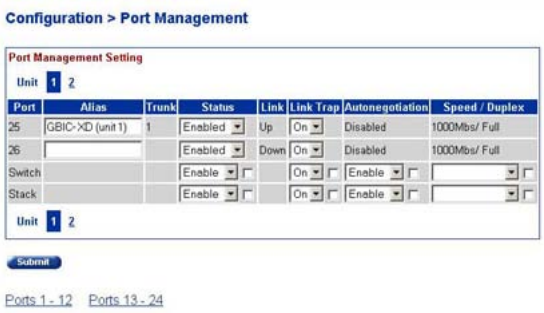

Figure 44: GBIC port (1) configuration

4. Configure the Call Server ELAN network interface on switch 2 (and 4) as shown in Figure 45: Call Server ELAN port configuration on page 162.

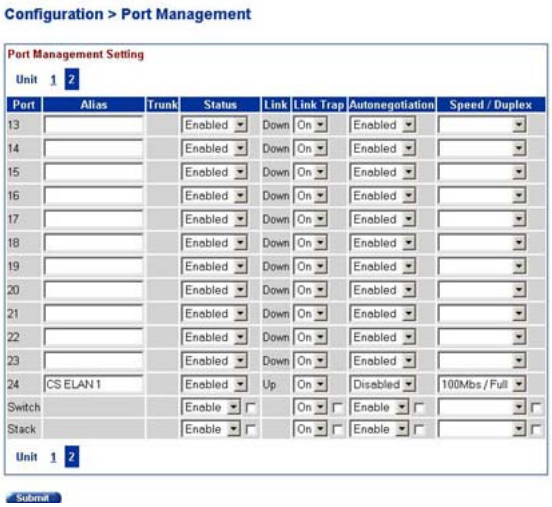

Figure 45: Call Server ELAN port configuration

5. Configure the GBIC network interface for switch 2 (and 4) as shown in Figure 46: GBIC port (2) configuration on page 162.

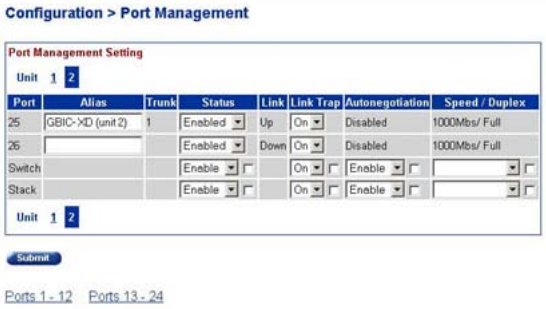

Figure 46: GBIC port (2) configuration

6. From the main menu, choose **Application > VLAN > VLAN Configuration**.
7. Configure the VLANs as shown in Figure 47: VLAN configuration on page 163.

Figure 47: VLAN configuration

8. From the main menu, choose **Application > VLAN > Port Configuration**.
9. Configure the VLAN for HSP on switch 1 (and 3) as shown in Figure 48: HSP VLAN configuration on page 163.

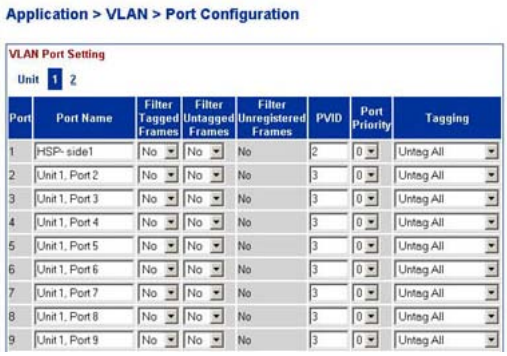

Figure 48: HSP VLAN configuration

10. Configure the GBIC VLAN network interfaces for switch 1 (and 3) as shown in Figure 49: GBIC (1) VLAN configuration on page 163.

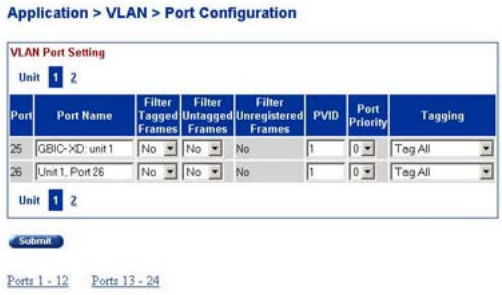

Figure 49: GBIC (1) VLAN configuration

11. Configure Call Server ELAN VLAN network interface on switch 2 (and 4) as shown in Figure 50: Call Server ELAN VLAN configuration on page 164.

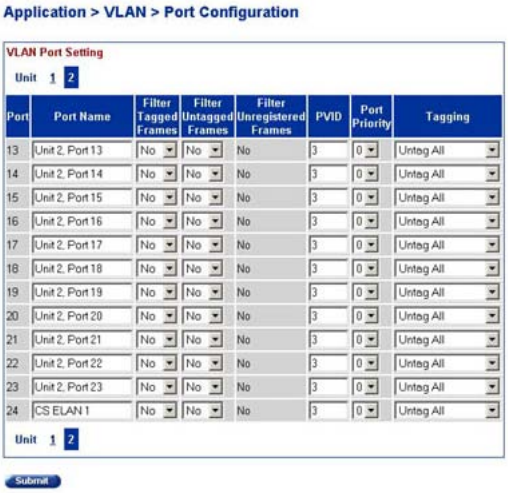

Figure 50: Call Server ELAN VLAN configuration

12. Configure the GBIC VLAN network interface for switch 2 (and 4) in the stack as shown in Figure 51: GBIC (2) VLAN configuration on page 164.

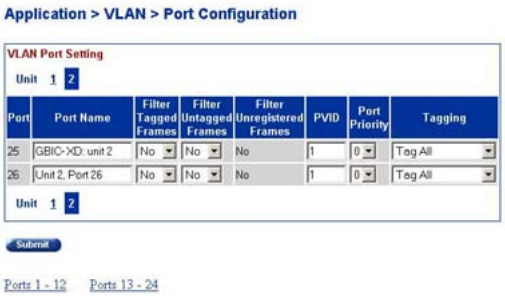

Figure 51: GBIC (2) VLAN configuration

13. From the main menu choose **Application > VLAN > VLAN Configuration**.

    The VLAN Configuration Web page appears For more information, see Figure 47: VLAN configuration on page 163.

14. Under **VLAN Creation**, in the **VLAN Type** drop-down list, choose **Port**.

15. Click **Create VLAN**.

    The VLAN Configuration: Port Based Web page appears.

16. Configure default VLAN port membership as shown in Figure 52: VLAN port membership configuration on page 165.

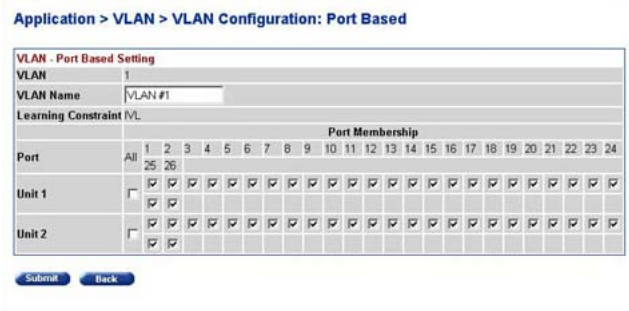

Figure 52: VLAN port membership configuration

17. Configure HSP VLAN port membership as shown in Figure 53: HSP VLAN port membership configuration on page 165.

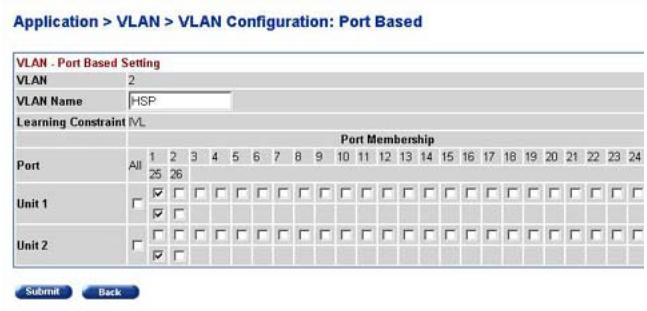

Figure 53: HSP VLAN port membership configuration

18. Configure ELAN VLAN port membership as shown in Figure 54: ELAN VLAN port membership configuration on page 165.

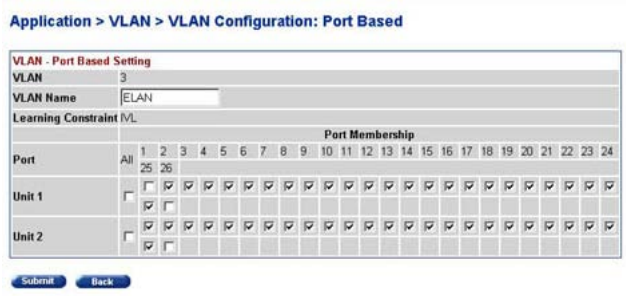

Figure 54: ELAN VLAN port membership configuration

19. From the main menu, choose **Application > MultiLink Trunk > Group**. Configure the MultiLink Trunk Group settings as shown in Figure 55: MultiLink Trunk Group settings configuration on page 166.

Application > MultiLink Trunk > Group

MultiLink Trunk Group Setting

Trunk	Trunk Members	STP Learning	Trunk Mode	Trunk Name	Trunk Status
1	Unit: 1 2 Port: 25 25	Fast	Basic	GBIC 1000Base->	Enabled
2	Unit: Port:	Normal	Basic	Trunk #2	Disabled
3	Unit: Port:	Normal	Basic	Trunk #3	Disabled
4	Unit: Port:	Normal	Basic	Trunk #4	Disabled
5	Unit: Port:	Normal	Basic	Trunk #5	Disabled
6	Unit: Port:	Normal	Basic	Trunk #6	Disabled

Figure 55: MultiLink Trunk Group settings configuration

Appendix B: Survivable SIP Media Gateway

Contents

This section contains information on the following topics:

- Survivable SIP Media Gateway task flow on page 167
- Upgrade a Survivable Media Gateway to a Survivable SIP Media Gateway on page 168
- Hardware Migration based on CP PM on page 171
- Hardware Migration based on CP DC on page 172
- Hardware Migration based on COTS on page 173
- Session Manager Network Connect Service on page 174
- SSMG call flows on page 179

Survivable SIP Media Gateway task flow

This section provides a high-level task flow for the installation or upgrade of a Survivable SIP Media Gateway. The task flow indicates the recommended sequence of events to follow when configuring a system and provides the book number that contains the detailed procedures required for the task.

For more information refer to the following books, which are referenced in the Survivable SIP Media Gateway task flow diagram:

- *Signaling Server IP Line Applications Fundamentals, NN43001-125*
- *Network Routing Service Fundamentals, NN43001-130*
- *Linux Platform Base and Applications Installation and Commissioning, NN43001-315*
- *Security Management Fundamentals, NN43001-604*
- *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*
- *Avaya Communication Server 1000E Software Upgrades, NN43041-458*

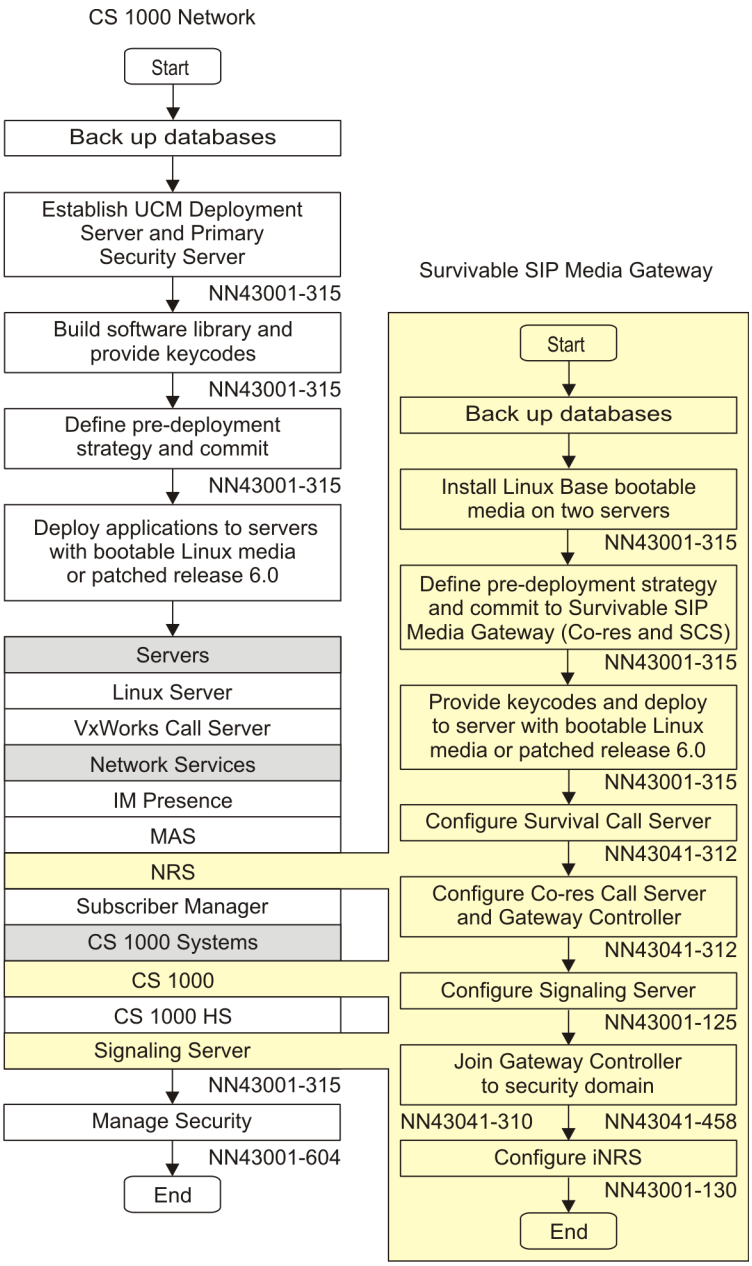

Figure 56: Survivable SIP Media Gateway task flow

Upgrade a Survivable Media Gateway to a Survivable SIP Media Gateway

Use this procedure to upgrade an existing Survivable Media Gateway (SMG) to a Survivable SIP Media Gateway.

The Media Gateway Controller of a Survivable Media Gateway registers to the Primary Call Server. In a Survivable SIP Media Gateway, the Media Gateway Controller registers to the resident SIP Media Gateway and uses SIP trunking to communicate with the Primary Call Server. The Survivable Call Server component serves as the Secondary Call Server and requires configuration similar to the SMG.

Note:

The Primary Call Server requires Package 404 (GRPRIM). The Secondary Call Server requires Package 405 (GRSEC). This procedure requires Avaya Communication Server 1000 Release 7.0 or later software.

Upgrading a Survivable Media Gateway to a Survivable SIP Media Gateway

1. Confirm that Avaya Communication Server 1000 software is installed on the Primary Call Server.

   - Configure GRPRIM Package 404.
   - Configure Signaling Servers and IPMGs.
   - Configure the other HA pairs for HS deployments (if required).

   For information about installing and configuring an Avaya CS 1000E Call Server, see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310*.

2. On the Primary Call Server, configure database replication.

   For information about configuring database replication, see Configuring Database Replication on the Primary Call Server on page 58.

3. Upgrade the Secondary Call Server on the existing Survivable Media Gateway to the Survivable Call Server.

   - Deploy Avaya Communication Server 1000 software to the Co-resident Call Server and Signaling Server applications software.
   - Deploy the NRS application.
   - Configure GRSEC Package 405.
   - Configure the GR Database Restoration policy. The Secondary Call Server database is replicated from the Primary Call Server using the existing GR database. Manually invoke the database download from the Primary Call Server.
   - Configure the Node ID.
   - Configure IPL/TPS.
   - Configure the SIP Gateway. (By default, the SIP Gateway registers with the Primary NRS. During a WAN outage, it can register with a local NRS or point directly to the Survivable Call Server.)
   - Configure the Local NRS for WAN outage.

4. Install a server card to function as the SIP Media Gateway. This server can be a CP PM or CP DC card.
   - Install Linux base on the server and register it to the Primary Security Domain.
   - Deploy Avaya Communication Server 1000 Co-resident Call Server and Signaling Server applications software.
   - Deploy the NRS application.
   - Configure the customer basic configuration and networking data.
   - Configure the Numbering Zone based on the overall Zone Dial Plan.
   - Configure the bandwidth zone for this SIP Media Gateway.
   - Configure digital trunks for PSTN access.
   - Configure ESN routing.
   - Configure TDM phones as required.
   - Configure the SIP Gateway component.
   - Configure the MGC data to register to the SIP Media Gateway.

   The SIP Media Gateway is not a GR Secondary Call Server, so it does not receive a replicated database from the Primary Call Server (PCS). The configuration for the SIP Media Gateway is independent of the PCS.
5. Install and configure the Signaling Servers.

   Note:

   There is an application Node ID for Secondary Call Servers associated with each Signaling Server node. This application Node ID must be the same as the Node ID used to define SIP signaling routes on the Primary Call Server.
6. Install the Primary, Alternate and Tertiary NRS.
   - The Tertiary NRS can be an actual NRS application running co-resident on the SCS signaling server or on the Survivable SIP MG Signaling Server.
   - The Survivable SIP MG Secondary Call Server and Survivable SIP Media Gateway SIP trunk applications can have each other's Node IP in their Tertiary NRS configuration section. This provides point-to-point operation without NRS registration.
7. On the Primary Call Server, use LD 43 to perform a data dump (EDD). This creates a current database for replication.
8. On the Primary Call Server, use LD 43 to specify the backup rule associated with each Secondary Call Server and replicate the database.
9. On each Secondary Call Server, log out from the system and wait until message SRPT4643 displays.

   This message notes the successful arrival of a new backup data file. Otherwise, after the manual restore and sysload, another automatic restore can occur (if ARSTR is defined) followed by an automatic sysload (if SYSLD is defined YES).

Note:

For any newly installed Secondary Call Server, you must use LD 117 to configure the corresponding Geographic Redundancy backup rule to allow automatic database restoration on the Secondary Call Server.

10. If required, install and configure IP Phones with S1 pointing to the secondary node TPS and S2 pointing to the Primary system node TPS.

Hardware Migration

There are a number of hardware options to configure a Survivable SIP Media Gateway. When migrating a Survivable Media Gateway to a Survivable SIP Media Gateway, additional server(s) will be needed depending on the option chosen. The list of Call Servers supporting the Survival Call Server portion of the Survivable SIP Media Gateway is the same as the list supported for Survivable Media Gateway operation. The following sections illustrate the possible hardware migration paths.

Hardware Migration based on CP PM

Three options are available:

1. The existing CP PM card serves as the Survivable Call Server. A new CP MG card is added to the gateway chassis. The existing MGC card is not used in this solution.
2. The existing CP PM card serves as the Survivable Call Server. An additional CP PM card is added to the gateway chassis, with the existing MGC card serving as the SIP Gateway.
3. Add a new CP PM card as the Survivable Call Server.. The existing CP PM and MGC cards serve as the basic SIP Media Gateway.

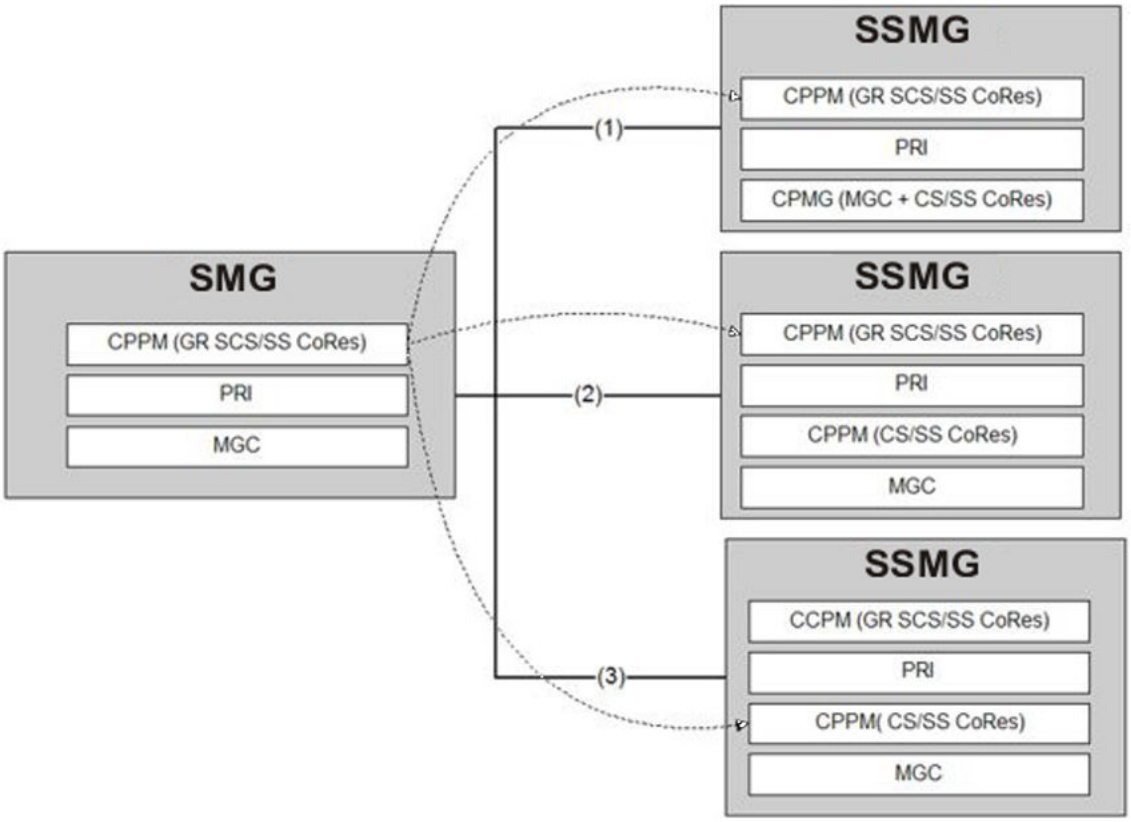

Figure 57: Migration based on CP PM

Hardware Migration based on CP DC

Three options are available:

1. A new CP DC card, serving as the Survivable Call Server, and a new CP MG card, serving as the basic SIP Media Gateway, are added to the gateway chassis.
2. The existing CP PM card serves as the Survivable Call Server. A new CP DC card is added to the gateway chassis to serve as the SIP Media Gateway with the existing MGC card.
3. The existing CP PM card serves as the SIP Media Gateway with the existing MGC. A new CP DC card, serving as the Survivable Call Server, is added to the gateway chassis.

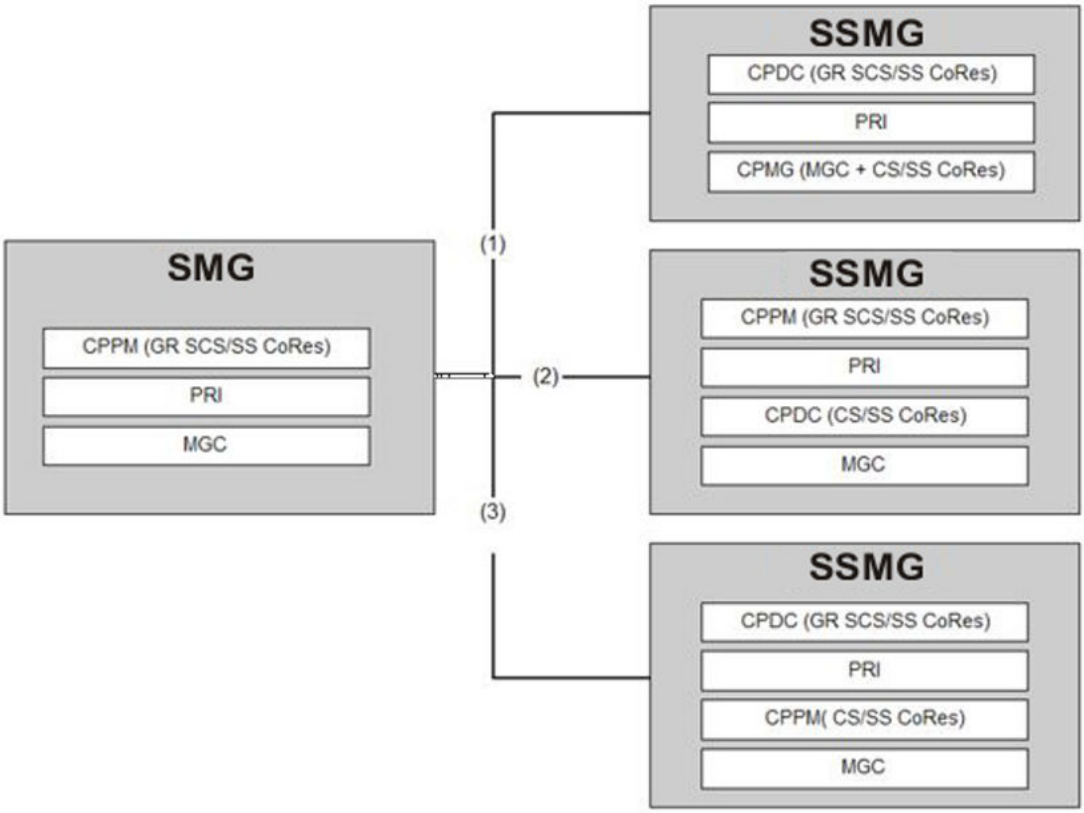

Figure 58: Migration based on CP DC

Hardware Migration based on COTS

Three options are available:

1. A new COTS serves as the Survivable Call Server. A new CP MG card serves as the SIP Media Gateway.
2. The existing CP PM card serves as the Survivable Call Server. A new COTS serves as the SIP Media Gateway with the existing MGC.
3. The existing CP PM card serves as the SIP Media Gateway with the existing MGC. A new COTS serves as the Survivable Call Server.

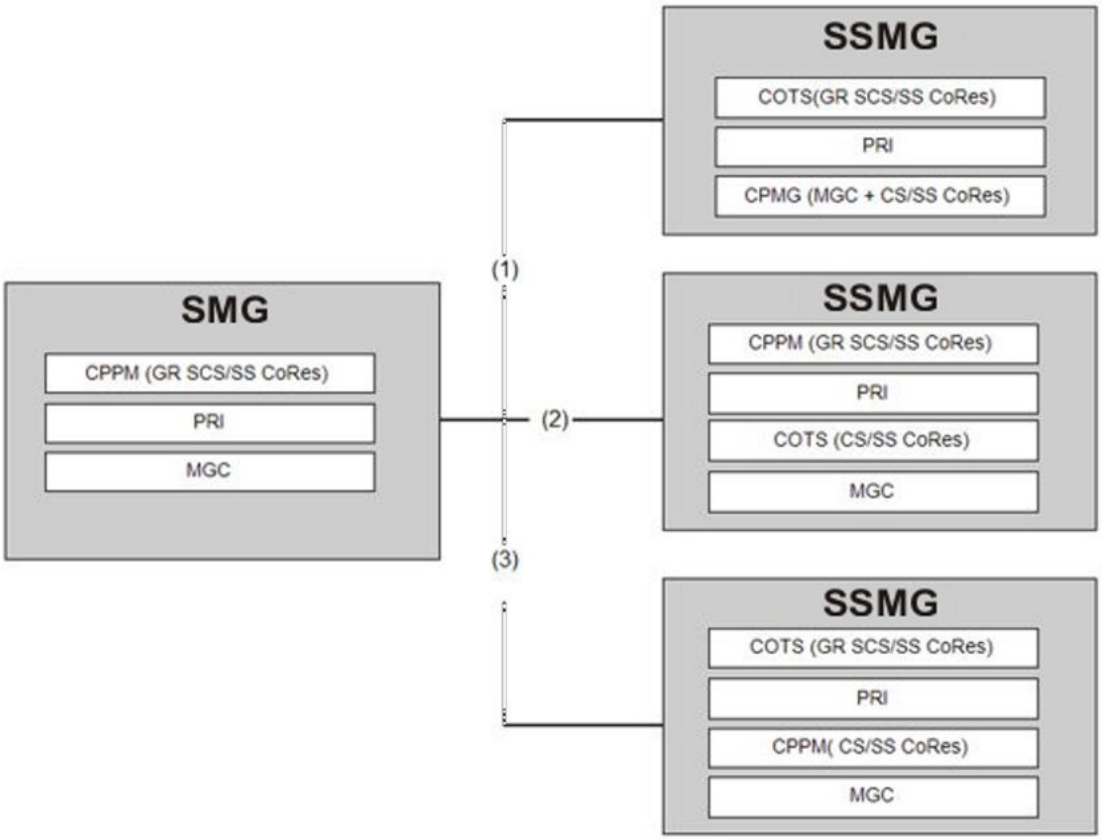

Figure 59: Migration based on COTS

Session Manager Network Connect Service

For information about migrating Communication Server 1000 systems to Session Manager, see the chapter Migrate to Avaya Aura ® Session Manager in *Network Routing Service Fundamentals, NN43001–130*.

NRS Proxy Users

Session Manager Network Connect Service (NCS) provides routing services to VoIP endpoints based on Avaya Communication Server 1000 (CS1000). This service is similar to the services provided by Nortel Network Routing Service (NRS). The NRS provides following three services:

- SIP Proxy Server (SPS): SPS performs two roles in CS1000-based configuration. These roles are SIP routing proxy and SIP registrar.
- Network Connect Server (NCS): NCS supports the centralized CS1000 solution for IP clients, that is, IP phones.
- H.323 Gatekeeper (GK): GK provides typical H.323 gatekeeper functions to H.323 endpoints in the solution.

Session Manager provides both the SIP Proxy Server (SPS) and Network Connect Server (NCS) functionality that are provided by Nortel NRS.

Session Manager provides the following information to each UNIStim phone:

- Primary signaling server
- Secondary signaling server
- Branch Office, Survivable Media Gateway or Survivable Remote Gateway

Session Manager NCS provides network redirection services for UNIStim phones for several CS1000 network features, such as geographic redundancy, virtual office, branch office, and Survivable Remote Gateway. You can define user patterns that are administered as a NRS Proxy User Rule in Session Manager, and associate user patterns with up to three CS 1000 Terminal Proxy Servers in priority order. Since a large number of users share the same server set, you can administer a range for the user pattern to match a large set of users.

Redirecting UNIStim Phones

About this task

In normal conditions, all UNIStim phones in the system should be configured with their Primary Connect Server, Primary CS 1000, pointing to the local Primary Session Manager and the Secondary Connect Server, Secondary CS 1000, pointing to the Secondary Session Manager.

Following are the administration steps:

Procedure

1. Point each SIP Signaling Gateway from the SIP Proxy Server (SPS) to the Session Manager. See CS1000 documentation.
2. Reconfigure NCS pointing to Session Manager. See CS1000 documentation.
3. Create Terminal Proxy Server instances for Primary signaling server and Secondary signaling server as per your system requirement.
4. Create Terminal Proxy Server instances for each Branch Office, Survivable Media Gateway, or Survivable Remote Gateway 50.
5. Create an NRS Proxy User Rule.

Creating Terminal Proxy Server instance

Procedure

1. On the System Manager console, in **Elements**, click **Inventory**.
2. From the left navigation pane, click **Inventory** > **Manage Elements**.
3. On the Manage Elements page, click **New** and select a “CS 1000 Terminal Proxy Server” entity instance.
4. On the New CS 1000 Terminal Proxy Server page, enter the following details:
   - In the **Name** field, enter the H323 ID value of the Main or Primary Signaling Server.
   - Enter the Node IP address of the Main or Primary Signaling Server.
5. Click **Commit**.

   When you add an application entity through Runtime Topology Service (RTS), it in turn starts a synchronization job in the background to bring all the relevant data from the application instances to the Communication System Management database. You can check the status of this synchronization job on the System Manager console by accessing **Scheduler** in **Services**.

Creating a NRS Proxy User Rule

Procedure

1. On the System Manager console, in **Elements**, click **Session Manager**.
2. To open the NRS Proxy Users page, click **Application Configuration** > **NRS Proxy Users** .
3. Click **New** and enter the appropriate information for the new NRS Proxy User Rule.
4. Click **Commit**.

   Note:

   Terminal Proxy Server should be administered using **Elements** > **Inventory** > **Manage Elements**.

Modifying NRS Proxy User Rule

Procedure

1. On the System Manager console, in **Elements**, click **Session Manager**.
2. To open the NRS Proxy Users page, click **Application Configuration** > **NRS Proxy Users**.
3. Select the NRS Proxy User Rule to be modified and click **Edit**.
4. On the NRS Proxy User Rule Editor page, modify the information and click **Commit** to save the changes.

   Note:

   Terminal Proxy Server must be administered using **Elements** > **Inventory** > **Manage Elements**.

Deleting NRS Proxy User rule

Procedure

1. On the System Manager console, in **Elements**, click **Session Manager**.
2. To open NRS Proxy Users page, click **Application Configuration** > **NRS Proxy Users**.
3. Select one or more NRS Proxy User rule from the list and click **Delete**.
4. On the Delete Confirmation screen, click **Delete** to delete the selected NRS Proxy User rules.

Delete Confirmation page field descriptions

Button	Description
Cancel	Cancels the deletion of the selected NRS Proxy User rule.
Delete	Deletes the selected NRS Proxy User rule.

NRS Proxy Users field descriptions

Button	Description
New	Adds a new entry.
Edit	Saves the changes of the modified entry.
Delete	Deletes the selected entry.

Name	Description
Pattern	User Pattern type.
Primary Terminal Proxy Server	Administered Terminal Proxy Server. This field is a required field.
Secondary Terminal Proxy Server	Administered Terminal Proxy Server.
Survivable Terminal Proxy Server	Administered Terminal Proxy Server.

NRS Proxy User Rule Editor field descriptions

Name	Description
Pattern	Specifies the User Pattern. This field is a required field. The prefix can include the numbers 0 to 9, and the hash (#), the hyphen (-), and the question mark (?) symbols. The prefix can have a length of up to 30 characters, but the first character must be numeric. Routing entries can be categorized in the following way: • Basic digits: Any number of digits can be used to make a simple entry. For example, 570 • Range of digits: Two basic digit entries can be used to build a range entry. The range will have a dash between the two ranges and each range will have the same number of digits. For example, 467 – 486 is valid but 467 – 4866 is invalid. • Length delimited: A basic digit entry can be terminated with the hash "#" symbol to

Name	Description
	force a specific length match. For example, 5703360#. • Wild card: The wild card digit character "?" can be appended to a basic digit entry. Multiple wild cards may be appended, but each wild card character matches only one digit. For example, 57033??.
Primary Terminal Proxy Server	Select an administered Terminal Proxy Server. This field is a required field.
Secondary Terminal Proxy Server	Select an administered Terminal Proxy Server.
Survivable Terminal Proxy Server	Select an administered Terminal Proxy Server.

Button	Description
Commit	Saves the changes made to the NRS Proxy User Rule.
Cancel	Cancels the changes made to the NRS Proxy User Rule.

SSMG call flow scenarios

For these scenarios:

- MO (Main Office) = PCS (Primary Call Server)
- GR (Geographic Redundancy) = ACS (Alternate Call Server)
- BO (Branch Office) = SSMG (Survivable SIP Media Gateway)

Normal call flow scenarios

In a normal call flow scenario, all BO IP phones are registered to MO with NCS redirection, all BO SIPL phones will be registered to MO, and TDM phones remain registered with BO. SIP Media Gateway SSG points to Session Managers.

a. BO TDM phone calls BO IP phone (registered with MO): call goes out from BO SIP Media Gateway SIP trunk to MO SIP trunk using SM and lands on BO IP phone (regular SIP trunk dialing plan is required to support this)

b. BO IP phone (registered with MO) calls BO TDM phone: call goes out from MO SIP trunk to BO SIP Media Gateway SIP trunk using SM and lands on BO TDM phone (regular SIP trunk dialing plan is required to support this)

c. BO PSTN trunk incoming call to BO IP phone (registered with MO): call comes from BO PSTN trunk, then goes out from BO SIP Media Gateway SIP trunk to MO SIP trunk using SM and then lands on BO IP phone (regular PSTN/SIP trunk dialing plan is required to support this)

d. BO IP phone (registered with MO) usually calls PSTN directly from MO if PSTN is available, no need to use BO PSTN trunk.

e. MO TDM phone calls BO IP phone (registered with MO): local call since BO IP phone is registered with MO, it has the same dialing plan with MO IP phone.

f. BO IP phone (registered with MO) calls MO TDM phone: local call since BO IP phone is registered with MO, it has the same dialing plan with MO IP phone.

g. MO IP phone calls BO TDM phone: the same as b.

h. BO TDM phone calls MO IP phone: the same as a.

i. MO TDM phone calls BO TDM phone: the same as g.

j. BO TDM phone calls MO TDM phone: the same as h.

k. BO PSTN trunk incoming call to BO TDM: call comes from PSTN and land locally (regular PSTN trunk dialing plan is required to support this)

l. BO TDM outgoing to BO PSTN trunk: call goes out to PSTN trunk directly (regular PSTN trunk dialing plan is required to support this)

m. SIP enabled CS 1000 (IP or TDM phone) calls BO IP phone (registered with MO): call goes out from CS1K SIP Media Gateway SIP trunk to MO SIP trunk using SM and lands on BO IP phone (regular SIP trunk dialing plan is required to support this)

n. SIP enabled CS 1000 calls BO TDM phone: call goes out from CS 1000 SIP Media Gateway SIP trunk to BO SIP trunk using SM and lands on BO IP phone (regular SIP trunk dialing plan is required to support this)

o. BO IP phone (registered with MO) calls SIP enabled CS 1000: call goes out from MO SIP trunk to CS 1000 SIP Media Gateway SIP trunk using SM and lands on IP / TDM phone (regular SIP trunk dialing plan is required to support this)

p. BO TDM phone calls SIP enabled CS 1000: call goes out from BO SIP trunk to CS 1000 SIP Media Gateway SIP trunk using SM and lands on IP / TDM phone (regular SIP trunk dialing plan is required to support this)

Branch is not available and cannot reach MO or GR (WAN Outage / SM not reachable)

All BO IP phones are registered to BO Soft Switch TPS, Soft Switch SSG and SIP Media Gateway SSG point to each other (Third), and TDM phones remain registered with BO.

a. BO TDM phone calls BO IP phone: call goes out from BO SIP Media Gateway SIP trunk to BO Soft Switch and lands on BO IP phone (regular SIP trunk dialing plan is required to support this)

b. BO IP phone calls BO TDM phone: call goes out from BO Soft Switch SIP trunk to BO SIP Media Gateway and then lands on BO TDM phone (regular SIP trunk dialing plan is required to support this)

c. BO PSTN trunk incoming call to BO IP phone: call comes from BO PSTN trunk, then goes out from BO SIP Media Gateway SIP trunk to BO Soft Switch SIP trunk and then lands on BO IP phone (regular PSTN/SIP trunk dialing plan is required to support this)

d. BO IP phone dial out to BO PSTN trunk: call goes out from BO Soft Switch SIP trunk to BO SIP Media Gateway SIP trunk, then out to PSTN trunk (regular PSTN/SIP trunk dialing plan is required to support this)

e. MO TDM phone calls BO IP phone: not reachable due to WAN outage except by PSTN alternate routing

f. BO IP phone calls MO TDM phone: not reachable due to WAN outage except by PSTN alternate routing: call goes out from BO Soft Switch SIP trunk to BO SIP Media Gateway and then connects to MO TDM phone using PSTN (regular PSTN/SIP trunk dialing plan is required to support this)

g. MO IP phone calls BO TDM phone: not reachable due to WAN outage except by PSTN alternate routing

h. BO TDM phone calls MO IP phone: not reachable due to WAN outage except by PSTN alternate routing

i. MO TDM phone calls BO TDM phone: not reachable due to WAN outage except by PSTN alternate routing

j. BO TDM phone calls MO TDM phone: not reachable due to WAN outage except by PSTN alternate routing

k. BO PSTN trunk incoming call to BO TDM: call comes from PSTN and land locally (regular PSTN trunk dialing plan is required to support this)

l. BO TDM outgoing to BO PSTN trunk: call goes out to PSTN trunk directly (regular PSTN trunk dialing plan is required to support this)

m. SIP enabled CS1000 calls BO IP phone (registered with MO): not reachable due to WAN outage except by PSTN alternate routing

n. SIP enabled CS1000 calls BO TDM Set: not reachable due to WAN outage except by PSTN alternate routing

o. BO IP phone (registered with MO) calls SIP enabled CS1000: not reachable due to WAN outage except by PSTN alternate routing

p. BO TDM phone calls SIP enabled CS1000 (IP Or TDM Set): not reachable due to WAN outage except by PSTN alternate routing

MO is not available and GR is still active (SMs are still reachable)

All BO IP phones are registered to GR, Soft Switch SSG and SIP Media Gateway SSG still point to SMs, and TDM phones remain registered with BO. All call flows are similar to the normal scenario except it is on GR instead of MO and a different dialing plan is required on SM.

MO and BO are still active but both SMs are unavailable (almost unlikely to happen)

This is not supported because all BO IP phones are still registered with MO. If both SMs are down, the Branch CS and IP phones do not know unless they are rebooted.

Appendix C: Controlled Load-sharing zones

Contents

This section contains information on the following topics:

Network bandwidth management zones on page 183

Configuring zone parameters at the backup site on page 185

Configuring zone parameters at the home site on page 190

Configuring zone-based digit manipulation on page 194

Network bandwidth management zones

An IP Peer network is divided into different bandwidth management zones, to which each IP Phone in the network is assigned. IP Phones in the same bandwidth management zone:

- share the same IP bandwidth management policies
- are geographically near each other
- are all in the same time zone
- are all in the same PSTN dialing plan

Each IP Phone is assigned to a zone during configuration. For dialing plan purposes, IP Phones in the same zone are treated identically.

Bandwidth management zones enable IP Phones that are located in separate geographic locations to have dialing plan behaviors that are localized to the telephone location rather than the Call Server location.

With the Controlled Load-sharing configuration, the backup system must be configured with one zone for its local IP Phones and with a second zone for the redundant IP Phones (multiple zones can also be configured for a system, to take account of, for example, different floors of a building). This allows the administrator to define a different numbering plan on the backup system for the home site IP Phones for local, long distance (optional), and emergency services calls. It also allows the administrator to configure the appropriate time display for the home site IP Phones when they are in a different time zone than the backup site.

When the home site IP Phones are registered on the backup system, zone configuration data enables the backup Call Server to modify the dialed digits for calls initiated from a home site

telephone. The NRS then provides the endpoint information to route the call to the appropriate destination.

Note:

Throughout this document, the term "zone" is defined as a bandwidth management zone, not an NRS zone.

Zone-based digit manipulation

Zone-based digit manipulation allows the Controlled Load-sharing configuration to provide users with seamless transition when system control is passed to the backup system following home system failure. Users can continue dialing local public numbers as normal even if system control has switched to a different NPA area.

To achieve this, the Zone Access Code Behavior (ZACB) and Zone Digit Prefix (ZDP) properties must be configured on the backup system for the redundant IP Phones. ZACB and ZDP are used to add digits to the digits dialed on the home site IP Phone. The resulting digit string is then used to route the call. The net effect is that redundant IP Phone users can continue to enter the same dialed digits and be routed appropriately under control of the backup system.

For example, if "1 87654321" is dialed, where "1" is the Access Code, then:

- when the IP Phone is registered at the home system, the call is routed based on the dialed digits.
- when the IP Phone is registered at the backup system, the digits undergo zone-based digit manipulation (such as inserting "101"), and the call is routed based on the new manipulated digit string (in this example "1 101 87654321").

Note:

Special considerations apply in the case where a single Access Code is used for both on-net and off-net calls, especially when UDP is used. Routing of on-net and off-net calls is normally different. The Call Server ESN Special Number provisioning and Gatekeeper Numbering Plan Entry provisioning should be used to provide this different routing.

In the case where a single Access Code is not shared, that is, where one Access Code is exclusively used for UDP on-net dialing, standard procedures should be used. Refer to *Avaya Dialing Plans Reference, NN43001-283*.

For a given home system, more than one zone can be defined at the backup system. Therefore, different home site IP Phones can receive different routing treatments at the backup site. The combination of zone-based digit manipulation and routing capabilities can be used to achieve many other routing outcomes for home site IP Phone calls at the backup system.

Comments? infodev@avaya.com

Zone configuration considerations

Do not configure Zone 0, the default zone, as a home or backup system zone. Network Bandwidth Management does not support zone 0. If zone 0 is configured as a system zone, the Bandwidth Management feature is not activated.

In the home and backup systems, configure available bandwidth and preferred strategy for a zone with LD 117 or Element Manager.

This section describes the configuration of zones on the backup system for the redundant home site IP Phones.

For more information about system configuration, see *IP Peer Networking Installation and Commissioning, NN43001-313*. Also see *Avaya Communication Server 1000E Installation and Commissioning, NN43041-310* or *Avaya Communication Server 1000M and Meridian 1 Small System Installation and Commissioning, NN43021-310*, as appropriate for the system.

To configure the zones at the home and backup system, perform the procedures in the following sections:

1. Configuring zone parameters at the backup site on page 185
2. Configuring zone parameters at the home site on page 190
3. Configuring zone-based digit manipulation on page 194

Configuring zone parameters at the backup site

This section describes how to configure zone parameters on the backup system to take into account the redundant home system IP Phones. The procedure is similar to an IP Peer Network configuration. For information about IP Peer Networking, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

Zone parameters are defined at the backup system in LD 117 (Configuring the home system zone on page 190. Time adjustments for zones are configured in LD 117 and defined relative to the time set in LD 2.

Note:

The time adjustment for the zone is required only on the backup system, to account for home IP Phones that are registered from a different time zone.

Warning:

Before and after an upgrade, perform a data dump (using LD 43 EDD or through Element Manager) on the Call Server to back up the existing data.

Configuring ESN and redundant IP Phone zones

1. On the backup system, Configure the Home Location Code (HLOC), and the Virtual Private Network Identifier (VPNI).

   Table 29: LD 15: Configure Customer Data Home Location Code and Virtual Private Network Identifier

Prompt	Response	Description
REQ:	CHG	Change existing data block.
TYPE:	NET	ISDN and ESN Networking options
CUST		Customer number
	0-99	Range for Avaya CS 1000M Systems and Avaya CS 1000E systems
...		
CLID	YES	Allow Calling Line Identification option
- ENTRY	xx	CLID entry to be configured
- - HLOC	100-9999999	Home location code (ESN) (3-7 digits)
ISDN	YES	Integrated Services Digital Network
- VPNI	(0)-16283	Virtual Private Network Identifier for Bandwidth Management Feature 0 or X = Disable feature 1-16383 = Enable feature <cr> = No Change

2. Configure Vacant Number Routing (VNR).

   VNR must be configured to ensure the necessary routing in the case of split-registration due to network connectivity failure.

   VNR is routed through the Virtual Trunk. This enables the NRS to centralize Numbering Plan definitions. To configure VNR, configure a Route List Index (RLI) with the Digit Manipulation Index (DMI) in LD 86 set to 0 (no digit manipulation required) on the Virtual Trunk route.

   Table 30: LD 15: Configure Vacant Number Routing

Prompt	Response	Description
REQ:	NEW CHG	Add new data or change existing data
TYPE:	NET	Configure networking
VNR	YES	Vacant Number Routing
- RLI	0-1999	Route List Index as defined in LD 86
- FLEN	1-(16)	Flexible length of digits expected
- CDPL	1-(10)	Flexible length of VNR CDP

Prompt	Response	Description
- UDPL	1-(19)	Flexible length of VNR LOC

3. On the backup system, create the home site zone.

   Configure the zone properties for IP telephony bandwidth management. Use LD 117 or Element Manager. For information about IP Peer networking, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

   Note:

   The home system zone number and zone bandwidth management parameters at the backup system must match the corresponding home system zone number and zone bandwidth management parameters at the home system.

   Important:

   Zone 0, the default zone, must not be configured as a system zone. Network Bandwidth Management does not support Zone 0. If Zone 0 is configured as a system zone, the Bandwidth Management feature is not activated.

Table 31: LD 117: Define zone properties on the backup system for the home site IP Phones

Command	Description
NEW ZONE <xxxxxxxx> [<intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> <zoneResourceType>]	
	Create a new zone with the following parameters: • xxx = 0-255xxxxx = 0-8000 zone number • intraZoneBandwidth = Intrazone available bandwidth (see Note Footnote.) 0-1 000 000 Kbit/s [1] • intraZoneStrategy = Intrazone preferred strategy (BQ for Best Quality or BB for Best Bandwidth) (see Note Footnote.) [2] • interZoneBandwidth = Interzone available bandwidth (see Note Footnote.) 0-1 000 000 Kbit/s • interZoneStrategy = Interzone preferred strategy (BQ for Best Quality or BB for Best Bandwidth) (see Note Footnote.) • zoneResourceType = zone resource type (shared or private), where - shared = Current default zone type. The IP Phones configured in shared zones use DSP resources configured

[1] If the Network Bandwidth Management feature is going to be used, the intraZoneBandwidth and interZoneBandwidth parameters must be configured to the actual available bandwidth.

[2] If the Network Bandwidth Management feature is going to be used, and the zone is going to be associated with a Virtual Trunk, the intraZoneStrategy and interZoneStrategy parameters must be configured to BQ.

Command	Description
	in shared zones. If all of the shared zones' gateway channels are used, the caller receives an overflow tone and the call is blocked. The order of channel selection for the gateway channels is: 1. channel from same zone as IP Phone is configured 2. any available channel from the shared zones channels - private = DSP channels configured in a private zone are used only by IP Phones that are also configured for that private zone. If more DSP resources than are available in the zone are required by these IP Phones, DSPs from other zones are used. However, IP Phones configured in shared zones cannot use the private zones' channels. The order of selection for the gateway channels is: 1. channel from same private zone as IP Phone is configured 2. any available channel from the pool of shared zones' channels

4. Define the zone parameters on the backup system for the redundant IP Phone zone. Use LD 117 or Element Manager. Refer to *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

Table 32: LD 117: Define zone parameters on the backup system for the home site IP Phones

Command	Description
CHG ZBRN <Zone> <yes|no>	
	Define a zone as a home system zone.
CHG ZDST <Zone> <yes|no> <StartMonth> <StartWeek> <StartDay> <StartHour> <EndMonth> <EndWeek> <EndDay> <EndHour>	
	If the home system observes Daylight Savings Time (DST), these parameters specify the start and end of DST. During DST, the clock automatically advances one hour forward.
CHG ZTDF <Zone> <TimeDifferencefromBackupSystem>	
	Specified in minutes, the time difference between the backup system and the home system when each is in a different time zone.
CHG ZDES <Zone> <ZoneDescription>	
	A name to render data display more meaningful.

5. Enable the features for the home site zone in LD 117.

Table 33: LD 117: Enable features on the backup system for home site zone

Command	Description
ENL ZBR <zone> ALL	Enables features for home system <zone>.

Element Manager zone configuration

From Element Manager, configure the home system-specific zone properties and time difference on the backup system.

The Zones window (see Figure 60: Zone configuration on the backup system on page 189 is the main window used for zone configuration in Element Manager. Select **System > IP Network > Zones** from the Element Managernavigator to open this window.

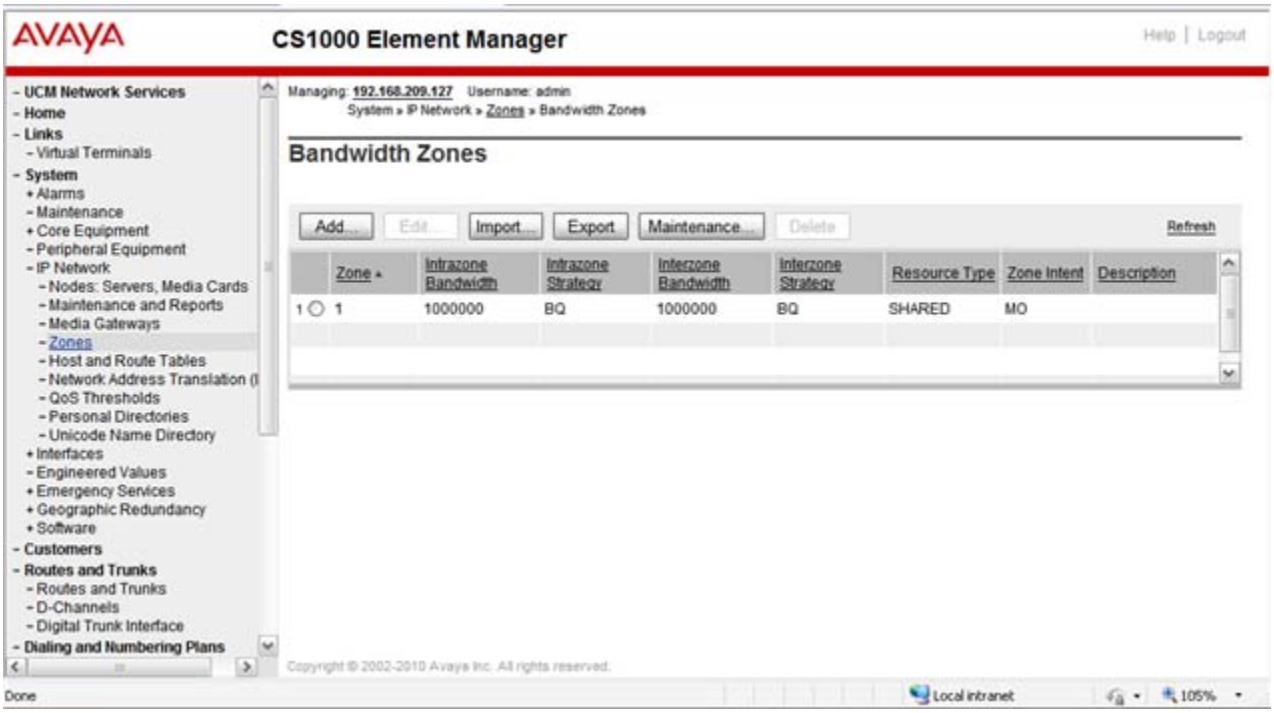

Figure 60: Zone configuration on the backup system

In the Zone List window, select the zone to be configured. The following properties can be configured:

- Basic Property and Bandwidth Management (see Figure 61: Zone Basic Property and Bandwidth Management on page 190
- Adaptive Network Bandwidth Management and CAC
- Alternate Routing for Calls between IP Stations
- Branch Office Dialing Plan and Access Codes

- Branch Office Emergency Service Information
- Branch Office Time Difference and Daylight Saving Time Property

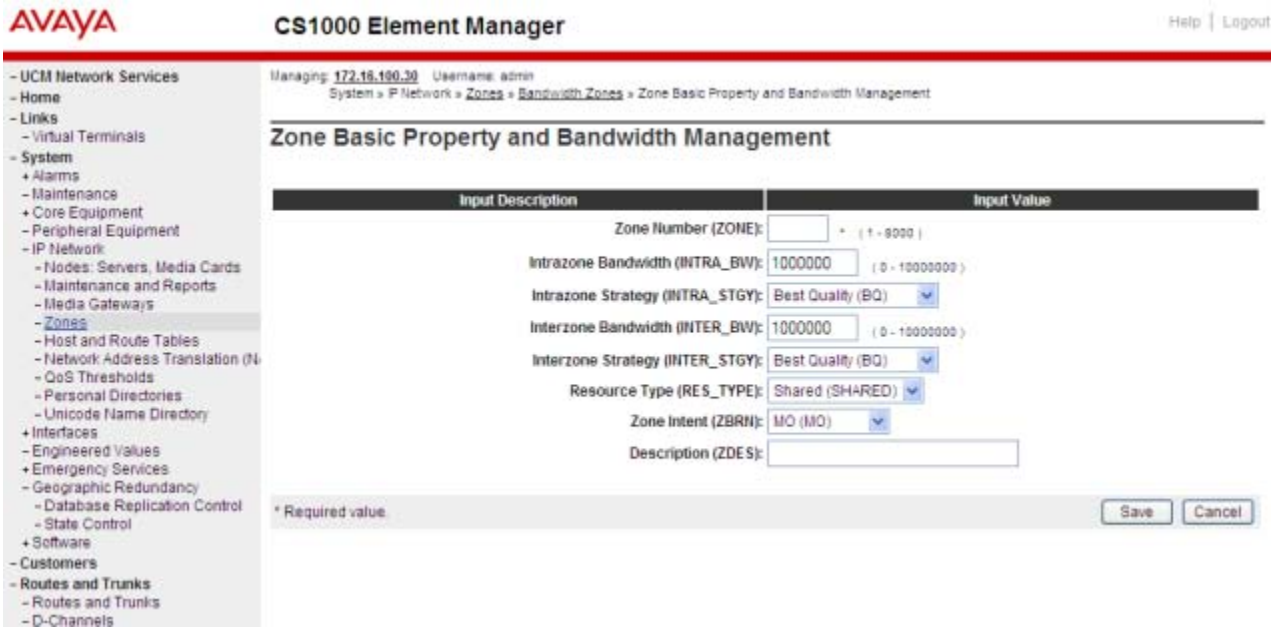

Figure 61: Zone Basic Property and Bandwidth Management

Configuring zone parameters at the home site

This section describes how to configure zone parameters on the home system to take into account the redundant IP Phones. The zones must be configured to match the zones configured on the backup system. The zones are defined in LD 117.

Warning:

Before and after an upgrade, perform a data dump (using LD 43 EDD or NRS Manager) on the Call Server to back up the existing data.

Configuring the home system zone

1. Configure the current date and time. For information about configuring the date and time, see *Avaya Software Input/Output Administration, NN43001-611*.

   Table 34: LD 2: Define system date

Command	Description
STAD dd mm yyyy hh mm ss	

Command	Description
	Configure the time and date: STAD DAY MONTH YEAR HOUR MINUTE SECOND

2. Configure the Home Location Code (HLOC) and Virtual Private Network Identifier (VPNI).

Table 35: LD 15: Configure Customer Data Home Location Code and Virtual Private Network Identifier.

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data
TYPE:	NET	ISDN and ESN Networking options
CUST	0-99	Customer number Range for Avaya CS 1000M systems and Avaya CS 1000E systems
...		
CLID	YES	Allow Calling Line Identification Option
- ENTRY	xx	CLID entry to be configured
- - HLOC	100-9999999	Home location code (ESN) (3-7 digits)
ISDN	YES	Integrated Services Digital Network
- VPNI	(0)-16283	Virtual Private Network Identifier for Bandwidth Management Feature 0 or X = disables feature 1-16383 = enables feature <cr> = no change

3. Configure VNR.

VNR must be configured to ensure the necessary routing in the case of split-registration due to network connectivity failure.

VNR is routed through the Virtual Trunk. This enables the NRS to centralize Numbering Plan definitions. To configure VNR, configure a RLI with the DMI in LD 86 set to 0 (no digit manipulation required) on the Virtual Trunk route.

Table 36: LD 15: Configure Vacant Number Routing

Prompt	Response	Description
REQ:	NEW CHG	Add new data or change existing data
TYPE:	NET	Configure networking
VNR	YES	Vacant Number Routing
- RLI	0-1999	Route List Index as defined in LD 86
- FLEN	1-(16)	Flexible length of digits expected
- CDPL	1-(10)	Flexible length of VNR CDP

Prompt	Response	Description
- UDPL	1-(19)	Flexible length of VNR LOC

4. Configure the zone properties for IP telephony bandwidth management. Use LD 117 or Element Manager (refer to Figure 62: Zone Basic Property and Bandwidth Management on page 193. At the home system, this zone is used only for bandwidth management purposes. It does not have any associated time zone or dialing plan properties.

Note:

The zone number and zone bandwidth management parameters at the home system must match the corresponding zone number and zone bandwidth management parameters at the backup system.

Important:

Zone 0, the default zone, must not be configured as a system zone. Network Bandwidth Management does not support zone 0. If zone 0 is configured as a system zone, the Network Bandwidth Management feature is not activated.

Table 37: LD 117: Define zone properties at the home system.

Command	Description
NEW ZONE <xxxxxxxx> [<intraZoneBandwidth> <intraZoneStrategy> <interZoneBandwidth> <interZoneStrategy> <zoneResourceType>]	
	Create a new zone with the following parameters: • xxx = 0-255xxxxx = 0-8000 zone number • intraZoneBandwidth = Intrazone available bandwidth 0-1 000 000 Kbit/s • intraZoneStrategy = Intrazone preferred strategy (BQ for Best Quality or BB for Best Bandwidth) • interZoneBandwidth = Interzone available bandwidth 0-1 000 000 Kbit/s • interZoneStrategy = Interzone preferred strategy (BQ for Best Quality or BB for Best Bandwidth) • zoneResourceType = zone resource type (shared or private), where - shared = Current default zone type. The IP Phones configured in shared zones use DSP resources configured in shared zones. If all of the shared zones' gateway channels are used, the caller receives an overflow tone and the call is blocked. The order of channel selection for the gateway channels is: 1. channel from same zone as IP Phone is configured

Command	Description
	2. any available channel from the shared zones' channels - private = DSP channels configured in a private zone are used only by IP Phones that are also configured for that private zone. If more DSP resources than are available in the zone are required by these IP Phones, DSPs from other zones are used. However, IP Phones configured in shared zones cannot use the private zones' channels. The order of selection for the gateway channels is: 1. channel from same private zone as IP Phone is configured 2. any available channel from the pool of shared zones' channels

Element Manager zone configuration on the home system

Figure 62: Zone Basic Property and Bandwidth Management on page 193 shows the only zone configuration screen required on the home system. It is an alternative to zone configuration using LD 117.

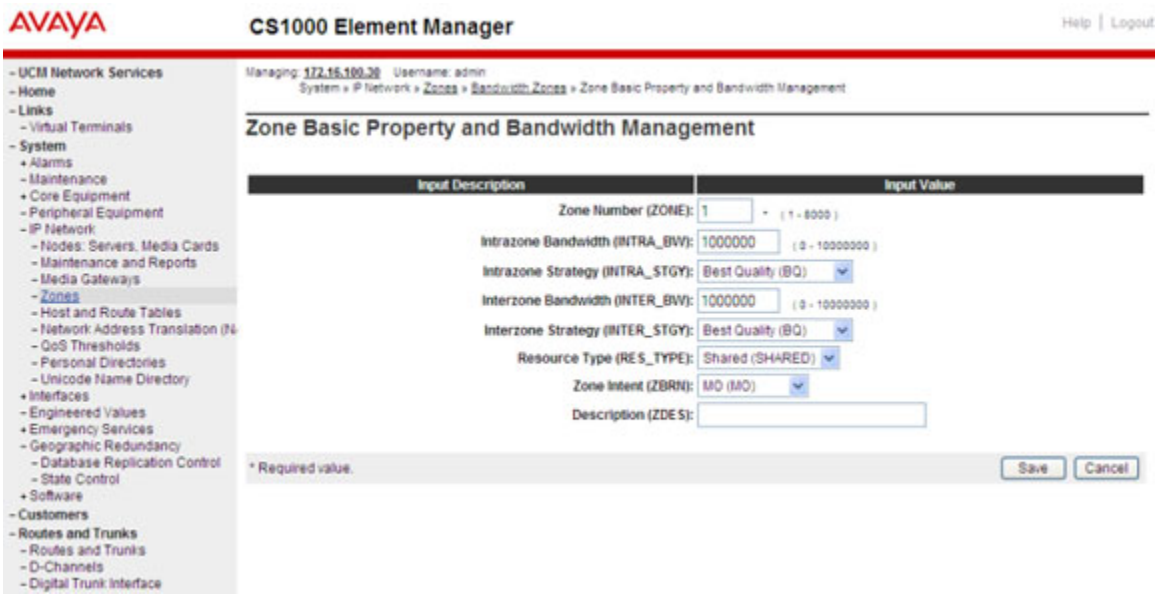

Figure 62: Zone Basic Property and Bandwidth Management

Add the Signaling Server name to the network NRS database using NRS Manager.

Note:

If the Signaling Server name is not added to the NRS database, the NRS rejects any registration request from the Signaling Server because its name is not in the ID list. The ID is case-sensitive.

For more information, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313* for the appropriate procedure.

Configuring zone-based digit manipulation

Perform the following steps to configure the dialing plan on the backup system to provide PSTN access to home site IP Phones:

1. Configure the ZACB property for the home system zone.
2. Configure the ZDP property for the home system zone.

You can perform these steps using overlays, as described in this section, or in Element Manager. For more information, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

Configuring the zone-based digit manipulation

1. Configure the ZACB property on the backup system for the home system zone.

   Table 38: LD 117: Define the zone Access Code handling for the home system zone.

Command	Description
CHG ZACB <zone> [ALL]|[<AC1|AC2> <AC1|AC2>]	
	Define the Access Codes used to modify local (or long distance) calls to force all calls local to the home site to be routed to the home site PSTN.

   The ZACB and ZDP properties are used to configure the digit manipulation behavior of the home system zone.

   The ZACB property specifies which calls undergo digit manipulation. The attribute can be configured in the following ways:

   - `CHG ZACB <zone>`

     In this configuration, dialing AC1 or AC2 does not trigger digit manipulation. Home system calls are treated exactly the same as those for backup system users.

   - `CHG ZACB <zone> ALL`

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In this configuration, calls dialed with AC1 and calls dialed with AC2 undergo zone-based digit manipulation.

For example, assume that AC1 = 1, AC2 = 2, and ZDP = 101. If a user dials "1 87654321", ZDP is inserted in the dialed digits to form a digit string of "1 101 87654321". If a user dials "2 87654321", ZDP is inserted in the dialed digits to form a digit string of "2 101 87654321".

- `CHG ZACB <zone> AC1 AC2`

  In this configuration, only calls dialed with AC1 undergo zone-based digit manipulation.

  For example, assume that AC1 = 1, AC2 = 2, and ZDP = 101. If a user dials "1 87654321", ZDP is inserted in the dialed digits to form a digit string of "2 101 87654321". If a user dials "2 87654321", zone-based digit manipulation does not occur and the digit string remains unchanged.

- `CHG ZACB <zone> AC2 AC2`

  In this configuration, only calls dialed with AC2 undergo zone-based digit manipulation.

  For example, assume that AC1 = 1, AC2 = 2, and ZDP = 101. If a user dials "1 87654321", zone-based digit manipulation does not occur and the digit string remains unchanged. If a user dials "2 87654321", ZDP is inserted in the dialed digits to form a digit string of "2 101 87654321".

Note:

As part of the ZACB configuration, the dialed Access Code can also be changed; so if AC2 is dialed, the Access Code can be changed to AC1, or vice versa. This provides more flexibility in the home system NARS configurations. Normally, there is no need to change the Access Code.

Note:

The Access Code dialed by the user is used internally by the Call Server. It is not sent as part of the outpulsed digits (to the NRS or to the trunks).

Note:

If a specified Access Code is used for both local and long distance dialing, then both types of calls receive the specified routing.

2. Configure the ZDP property for the home system zone at the backup system. For more information, see *Avaya IP Peer Networking Installation and Commissioning, NN43001-313*.

Table 39: LD 117: Define the zone digit manipulation on the backup system for the home site zone

Command	Description
CHG ZDP <zone> <DialingCode1> <DialingCode2> <DialingCode3>	

Command	Description
	Define the dialing plan for the home system zone, where DialingCode1, DialingCode2, and DialingCode3 are inserted into the dialed digits between the Access Code and the remainder of the dialed number.

The ZDP and ZACB (1 on page 194) properties are used to configure the digit manipulation behavior of the home system zone.

The ZDP property is inserted between the Access Code specified in the ZACB command and the dialed digits. This zone-based digit manipulation allows the backup system Call Server and the network NRS to distinguish the home site IP Phone calls from the backup site IP Phone calls, and route them accordingly. The digit manipulation occurs before any digit processing in the backup system Call Server or NRS.

Note:

If DialingCode1, DialingCode2, or DialingCode3 are already present in the dialed digits, then they are not re-inserted.

The Access Code ("1") is not included in the digit string that is sent to the NRS. The NRS recognizes "101" at the front of the digit string and routes the call to the destination.

Configuration example for PSTN resources

IP Phones registered on the backup system can be grouped into one of two categories:

- those physically located at the backup site and, therefore, configured with the backup site dialing plan
- those physically located at the home site and, therefore, configured with a dialing plan for the home site zone

Customer data must first be configured to recognize numbers that are local to each location (a standard NARS configuration issue). This example specifically focuses on the additional changes necessary to physically enable a home system telephone, registered with the backup system, to reach the PSTN at the home site.

Note:

Assume that the home system and backup system are configured with local numbers, such as 555-1212 or 967-1111.

Table 40: Example dialing string, area codes, and Access Codes on page 197 uses the following configuration at the backup system for home system telephones to reach the PSTN.

Table 40: Example dialing string, area codes, and Access Codes

	At the backup system node	At the home system node
Local dialing string	Local calls use 7-digit dialing.	Local calls use 7-digit dialing.
Area code (NPA)	The NPA is 613.	The NPA is 506.
Country code	The home system Node Country Code is 1.	The backup system Node Country Code is 1.
NARS configuration	Local calls use AC2, which is "9". Long-distance calls use AC1, which is "6".	Local calls use AC2, which is "9". Long distance calls use AC1, which is "6".
The Public National (E.164) entry points to...	"506" points to home system node.	"613" points to backup system node.

At the backup system, a zone must be configured for the home site IP Phones. In the definition of the home site zone, the ZACB and ZDP properties must be configured to insert the home site NPA into the dialed digits.

If a local backup system telephone goes off-hook and dials "9 555-1212", the Call Server assumes the user intends to reach the number 555-1212 in the local NPA. The fully-qualified number (E.164) is 1-613-555-1212.

If a home site IP Phone user goes off-hook and dials "9 555-1212" and the ZDP property is configured, the Call Server directs the user to the number 555-1212 in the NPA local to the home user. The fully-qualified number (E.164) is 1-506-555-1212.

Note:

You must change bandwidth zones parameters like VPNI and ZDP only during maintenance. If you change bandwidth zone parameters during H323/SIP calls it will lead to various issues.