Nortel Communication Server 1000

Main Office Configuration Guide for SRG 50

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Contents

New in this release

The following sections detail what's new in *CS 1000 Main Office Configuration for SRG 50* (NN43001-307) for Communication Server 1000 Release 6.0.

Features

This document contains no new features for CS 1000 Release 6.0.

Other

There are no other changes.

Revision history

May 2009	Standard 04.01. This document is up-issued to support CS 1000 Release 6.0.
July 2008	Standard 03.02. This document is up-issued to reflect changes in technical content. Sections relating to Bandwidth Management have been moved to *Converging the Data Network with VoIP Fundamentals* (NN43001-260).
December 2007	Standard 03.01. This document is up-issued to support CS 1000 Release 5.5 for SRG 50 Release 3.0.
November 2007	Standard 02.02. This document is up-issued to support CS 1000 Release 5.0 for SRG 50 Release 3.0. This document includes SIP Trunks configuration at the main office.
August 2007	Standard 02.01. This document is up-issued to support CS 1000 Release 5.0 for SRG 50 Release 3.0.
June 2007	Standard 01.02. This document is up-issued to remove the Nortel Networks Confidential statement.

May 2007	Standard 01.01. This document is up-issued to support Communication Server 1000 Release 5.0. This document contains information previously contained in the following legacy document, now retired: (553-3001-207). This document is up-issued to include updated information due to CR Q01587820.
October 2006	Standard 3.00. This document is up-issued to support SRG 50 Release 2.0 for CS 1000 Release 4.5.
January 2006	Standard 2.00. This document is up-issued for CR Q01202736, with information on reconfiguring Call Server alarm notification levels if necessary when configuring Adaptive Network Bandwidth Management.
August 2005	Standard 1.00. This document is a new document to support Communication Server 1000 Release 4.5.

Subject

This document *CS 1000 Main Office Configuration for SRG 50* (NN43001-307) describes the software for CS 1000 Release 6.0. Information in this document complements information found in documents in the Communication Server 1000 documentation suite. For information about how to configure the SRG 50, see *SRG 50 Configuration Guide* (NN40140-500) at www.nortel.com. Select Support & Training > Technical Documentation > Communication Servers > BCM.

Intended audiences

This document is intended for individuals responsible for configuring the main office for Survivable Remote Gateway for organizations using CS 1000 systems.

Conventions

Terminology

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

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

Related information

This section lists information sources that relate to this document.

NTPs

The following NTPs are referenced in this document:

- *Converging the Data Network with VoIP Fundamentals* (NN43001-260)
- *Electronic Switched Network Reference—Signaling and Transmission* (NN43001-280)
- *Dialing Plans Reference* (NN43001-283)
- *Signaling Server IP Line Applications Fundamentals* (NN43001-125)
- *IP Peer Networking Installation and Commissioning* (NN43001-313)
- *Branch Office Installation and Commissioning* (NN43001-314)
- *Telephony Manager 4.0 System Administration* (NN43050-601)
- *Software Input Output Reference-Administration* (NN43001-611)
- *Emergency Service Access Fundamentals* (NN43001-613)
- *Element Manager System Reference - Administration* (NN43001-632)
- *ISDN Primary Rate Interface Fundamentals* (NN43001-569)
- *Basic Network Feature Fundamentals* (NN43001-579)
- *Communication Server 1000M and Meridian 1 Large System Planning and Engineering* (NN43021-220)
- *Communication Server 1000E Planning and Engineering* (NN43041-220)
- *Software Input Output Reference - Maintenance* (NN43001-711)
- *SRG 50 Configuration Guide* (NN40140-500)

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Description

Contents

This section contains information about the following topics:

Survivable Remote Gateway

The Survivable Remote Gateway (SRG) extends the desktop feature and user interface of the CS 1000 to remote IP branch office users and gives them full access to the same applications as the main site. CallPilot, Contact Center Management Server (CCMS), and other central applications are shared by remote users to deliver state-of-the-art features and functionality to small remote offices.

SRG 50 Release 2.0 provides the following:

- extends the supported number of survivable IP users from 32 to 80
- extends support for the IP Phone 1120E, IP Phone 1140E, IP Audio Conference Phone 2033, and WLAN 2212

  See "Supported IP Phones" (page 15) for a complete list of supported IP Phones.
- supports H.323 and SIP Trunking to the CS 1000 main office
- supports analog devices, such as fax machines and terminals but are limited in number and limited to basic access

SRG 50 Release 3.0 provides the following:

- extends support for the IP Phone 1110, IP Phone 1210, IP Phone 1220, and IP Phone 1230

  See “Supported IP Phones” (page 15) for a complete list of supported IP Phones.
- evolves the SIP trunk to support a standard SIP Trunk interface
- supports On-site Notification

Table 1
Supported software at the branch office

IP branch office solution	Survivable users	Server support	Feature description
SRG 1.0	up to 90	Succession 3.0 CS 1000 Release 4.0 CS 1000 Release 4.5	VoIP and Application Gateway Local Mode = Basic telephony features
SRG 50 Release 1.0	up to 32	Succession 3.0 CS 1000 Release 4.0 CS 1000 Release 4.5	VoIP and Application Gateway Local Mode = Basic telephony features A more cost effective small branch office solution. Provides H.323 trunking. For more information, see *CS 1000 Main Office Configuration Guide for SRG 50* (553-3001-207) .
SRG 200/400 Release 1.5	up to 90	Succession 3.0 CS 1000 Release 4.0 CS 1000 Release 4.5 CS 1000 Release 5.0 CS 1000 Release 5.5	VoIP and Application Gateway Local Mode = Basic telephony features Feature Parity with SRG 50, new OS, and extended IP Phone support. Provides H.323 trunking. For more information, see *Main Office Configuration Guide for SRG 200/400 Release 1.5* (NN43001-308) .

IP branch office solution	Survivable users	Server support	Feature description
SRG 50 Release 2.0	up to 80	Succession 3.0 CS 1000 Release 4.0 CS 1000 Release 4.5 CS 1000 Release 5.0 CS 1000 Release 5.5	VoIP and Application Gateway Local Mode = Basic telephony features Extends IP Phone support and survivable IP users from 32 to 80. Provides H.323 and SIP trunking. For more information, see *CS 1000 Main Office Configuration Guide for SRG 50* (553-3001-207) .
SRG 50 Release 3.0	up to 80	CS 1000 Release 5.0 CS 1000 Release 5.5 CS 1000 Release 6.0	VoIP and Application Gateway Local Mode = Basic telephony features Extends IP Phone support to include the IP Phone 1110. Supports On Site Notification for E-911 calls. Provides H.323 and SIP trunking. CS 1000 Release 5.5 extends support to include IP Phone 1210, IP Phone 1220, and IP Phone 1230.
MG 1000B	up to 400	Succession 3.0 CS 1000 Release 4.0 CS 1000 Release 4.5 CS 1000 Release 5.0 CS 1000 Release 5.5 CS 1000 Release 6.0	100% CS 1000 feature and application redundancy in survivable mode. Designed and positioned for larger IP branch offices. Provides H.323 and SIP trunking.
MG 1000E	up to 400	CS 1000 Release 5.0 CS 1000 Release 5.5 CS 1000 Release 6.0	Provides survivability with the addition of Call Processor Pentium Mobile (CP PM).

The SRG is implemented on a BCM 50 platform and is connected to a CS 1000 at the main office through Virtual Trunks over a reliable IP WAN access facility. This configuration allows the call processing for the IP

Phones at the SRG site to be centralized at the main office. The Call Server at the main office provides the call processing for the IP Phones registered to both the main office and branch offices. The SRG provides call processing functionality to phones in local mode and local analog devices. The SRG supports business continuity and call failover through digital and analog trunk access to the local Public Switched Telephone Network (PSTN).

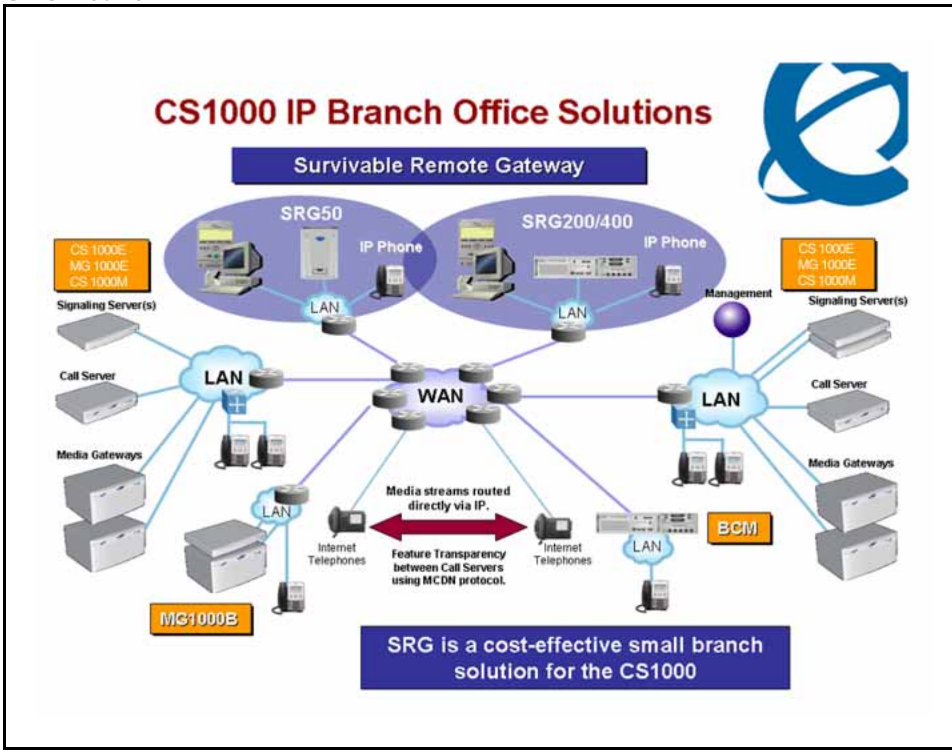
In order for devices in the CS 1000 network to access analog devices at the SRG or to access the PSTN at the SRG, virtual trunks are used over the LAN/WAN.

If the main office fails to function, or if there is a network/WAN outage, the SRG automatically switches to Local mode and provides basic telephony service to the phones located at the branch office. This enables the IP Phones to survive the outage between the branch office and the main office.

To ensure proper operation of the SRG solution it must be configured to support a common dialing plan with the CS 1000 main office. Any other configuration is not guaranteed to work reliably. Since the Call Server and the SRG handle dialing slightly differently, ensure that any settings you use for the main office that need to interact with the SRG, can be accommodated by the SRG call processing.

Figure 1 "SRG network" (page 13) shows the networking among the main office, SRG, and IP Phones.

Figure 1
SRG network



Main office hardware description

The main office must be one of the following systems:

- CS 1000E
- CS 1000M Cabinet
- CS 1000M Chassis
- CS 1000M HG
- CS 1000M SG
- CS 1000M MG

Throughout this document, references to CS 1000 systems encompass all CS 1000 system types.

The diagrams throughout this documentation show a CS 1000E main office. All of the systems appearing in the list perform identical main office functions as far as the SRG is concerned. For information about the SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Signaling Server

The following Signaling Servers are available for CS 1000 Release 6.0:

- Common Processor Pentium Mobile (CP PM)
- Common Processor Pentium Mobile (CP PM) Co-resident
- HP-DL320-G4
- HP-DL320-G5
- IBM-X306m
- IBM-X3350
- Dell R300

The Signaling Server is required at the main office only. It provides the following functions:

- Terminal Proxy Server (TPS)

  The TPS provides a connection from the IP Phones to the Call Server and a connection from a Virtual Trunk to the Call Server.
- H.323 Gateway (Virtual Trunk)
- SIP Gateway (Virtual Trunk)
- CS 1000 Element Manager Web Server and Network Routing Service (NRS)
- NRS, consisting of:
  - SIP Redirect Server NRS
  - H.323 Gatekeeper
  - Network Connection Service (NCS)
- Personal Directory

A second Signaling Server can be used to provide redundancy in the case of a failure in the primary Signaling Server at the main office.

A similar function to the Signaling Server is used at the SRG when the phones are in local mode.

The Signaling Server supports en bloc signaling which is standard on the Signaling Server.

For more information about the Signaling Server, see *Signaling Server IP Line Applications Fundamentals* (NN43001-125) . For more information about H.323 and overlap signaling, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

Network Routing Service

The Network Routing Service (NRS) application provides network-based routing, combining the following into a single application:

- **H.323 Gatekeeper**—provides central dialing plan management and routing for H.323-based endpoints and gateways.
- **SIP Redirect Server NRS**—provides central dialing plan management and routing for SIP-based endpoints and gateways. SIP Trunks are used for Voice packet traffic alone.
- **NRS Database**—stores the central dialing plan in XML format for the H.323 Gatekeeper, and the SIP Redirect Server. The H.323 Gatekeeper and the SIP Redirect Server accesses this common endpoint and gateway database.
- **Network Connect Server (NCS)**—used only for Media Gateway Controller (MGC) based MG 1000B, SRG, Geographic Redundancy, and Network-wide Virtual Office solutions. The NCS allows the Line TPS (LTPS) to query the NRS.
- **NRS Manager web interface**—the NRS provides its own web interface to configure the H.323 Gatekeeper, SIP Redirect Server, and the NCS.

The NRS application provides routing services to H.323 devices and SIP-compliant devices. The H.323 Gatekeeper can be configured to support H.323 routing services, while the SIP Redirect Server NRS can be configured to support SIP routing services. The H.323 Gatekeeper and the SIP Redirect Server NRS can reside on the same Signaling Server.

Each system in an IP Peer network must register to the NRS. The NRS software identifies the IP addresses of systems based on the network-wide numbering plan. NRS registration eliminates the need for manual configuration of IP addresses and numbering plan information at every site.

When configuring the NRS it is necessary to enable the NCS. Ensure that the check box “Network Connection Server enabled” is checked in the NRS configuration window of CS 1000 Element Manager.

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

Supported IP Phones

Table 2 "IP Phone support " (page 16) shows the supported IP Phones for each software release.

Table 2
IP Phone support

IP Phone	Release 1.0	Release 2.0	Release 3.0
IP Phone 2001	Supported	Supported	Supported
IP Phone 2002	Supported	Supported	Supported
IP Phone 2004	Supported	Supported	Supported
IP Phone 2007	Supported	Supported	Supported
IP Audio Conference Phone 2033	Supported	Supported	Supported
IP Phone 1210	Not supported	Not supported	Supported
IP Phone 1220	Not supported	Not supported	Supported
IP Phone 1230	Not supported	Not supported	Supported
IP Softphone 2050	Supported	Supported	Supported
IP Phone 1110	Not supported	Not supported	Supported
IP Phone 1120E	Not supported	Supported	Supported
IP Phone 1140E	Not supported	Supported	Supported
IP Phone 1150E	Not supported	Not supported	Not supported
Mobile Voice Client (MVC) 2050	Supported	Supported	Supported
Analog (500/2500-type) telephones	Supported	Supported	Supported
WLAN Handset 2210	Supported	Supported	Supported
WLAN Handset 2211	Supported	Supported	Supported
WLAN Handset 2212	Not supported	Supported	Supported
WLAN Handset 6120	Not supported	Not supported	Not supported
WLAN Handset 6140	Not supported	Not supported	Not supported

Throughout this document, the IP Phones are referred to collectively as IP Phones.

Main office requirements

The branch office running SRG Release 3.0 requires the following at the main office:

- CS 1000 hardware, running CS 1000 Release 4.0, CS 1000 Release 4.5, CS 1000 Release 5.0, CS 1000 Release 5.5, or CS 1000 Release 6.0.
- Configure at least one of the following packages for IP Peer Networking:

- — H.323 Virtual Trunk (H323_VTRK) package 399
- — SIP Gateway and Converged Desktop Package (SIP) package 406

- The main office must have a software Service Level of 2 or higher to work with the branch office.
- Ensure that you have ordered enough IP user and Virtual Trunk licenses at the main office to support the SRG 50 or the capacity of your branch office.

  The two different IP user licenses at the main office are:

  — Basic IP License for the IP Phone 2001, IP Audio Conference Phone 2033, IP Phone 1110, and IP Phone 1210

  — IP User License for the IP Phone 2002, IP Phone 2004, IP Phone 2007, IP Phone 1220, IP Phone 1230, IP Phone 1120E, IP Phone 1140E, IP Softphone 2050, Mobile Voice Client (MVC) 2050, WLAN Handset 2210, WLAN Handset 2211, and WLAN Handset 2212

The main office requires the following software packages to support the specified Basic Network features. See *Basic Network Feature Fundamentals* (NN43001-579) for more information about these features.

- Network Call Back Queuing (MCBQ) package 38. This package is required for SRG IP Phones to invoke any queuing feature or ringback when free.
- Network Speed Call (NSC) package 39. This package is required for SRG IP Phones to invoke the Network Speed Call feature.

The main office requires the following software packages to support the specified ISDN Primary Rate Interface features. See *ISDN Primary Rate Interface Fundamentals* (NN43001-569) for more information about these features.

- Network Attendant Service (NAS) package 159. This package is required for analog (500/2500-type) telephones in the branch office to access attendant services when the attendant is configured on the main office.
- Network Message Services (NMS) package 175. This package is required for analog (500/2500-type) telephones in the branch office to share the voice mail system in the main office. For any configurations using centralized CallPilot on the main office with one or more branch offices in separate time zones, the NMS package is required at the main office for the branch IP Phones.

Optional features to enhance SRG functionality

- Network Alternate Route Selection (NARS) package 58. See *Basic Network Feature Fundamentals* (NN43001-579) .
- Overlap Signaling (OVLP) package 184. This package is optional; it is required for overlap signaling. It is packaged with H.323 Virtual Trunk (H323_VTRK) package 399.
- Emergency Services Access (ESA) package 329. This package is optional; it is required only to receive 911/ESA features in North American and some Caribbean and Latin American (CALA) markets. See *Emergency Service Access Fundamentals* (NN43001-613) .
- Virtual Office (VIRTUAL_OFFICE) package 382. This package is optional; it is required only for Virtual Office functionality.
- Network Signaling (NSIG) package 37. This package is optional for SRG IP Phones to access set-based Network Class of Service (NCOS) features.
- Adaptive Network Bandwidth Management package 407.
- Alternative Call Routing for Network Bandwidth Management.

For software and hardware requirements for SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Normal Mode and Local Mode overview

Normal Mode and Local Mode overview provides a description of the following sections:

- Normal Mode
- Local Mode
- Survivability
- Recovery to Normal Mode
- Local Mode operation
- Virtual trunks

Normal Mode

IP Phones that are physically located at the SRG but are registered with the main office are operating in Normal Mode. In Normal Mode, the main office provides centralized call processing to all applications transparently to all IP Phones at the Branch Office. All IP Phones at the Branch, in Normal Mode, are registered to the main office TPS and are controlled by the Call Server at the main office.

Users of the SRG IP Phones receive the features, applications, key layout, and tones of the main office Call Server. This provides feature and application transparency between the branch office and the main office.

Local Mode

Users at the branch office may be in Local Mode, or survivable mode for two different reasons:

1. IP Phone has just booted up.
2. IP Phone cannot communicate to the main office because of a WAN failure or a failure of the main office components.

> **ATTENTION**
> When a telephone or trunk in the main office calls an SRG IP Phone that has switched to Local Mode due to WAN failure, the call is treated according to the main office call redirection configuration (such as forwarding to voice mail or continuous ringback).

In the event that the IP Phones at the branch office lose the connection to the main office CS 1000 call server for any reason (WAN failure, main office call server failure, main office Signaling Server failure), the SRG 50 reverts to Local Mode automatically. Essentially, when VoIP connectivity is lost, each IP Phone loses its Reliable UDP (RUDP) connection with the main office Terminal Proxy Server (TPS). The IP Phones at the branch office reboot and reregister to the SRG 50, placing them in Local Mode.

After this occurs, the IP Phones displays an indication on the display area that the set is in Local Mode of operation. This display is configurable by installers to meet local language and usage norms.

In Local Mode, the IP users connected at the branch office are under the control of the SRG 50 call services. As such, the normal main office call server features are not available. The SRG 50 offers a basic feature set when in Local Mode which allows IP Phones to continue to make and receive calls internally within the branch office and over the provisioned local PSTN interfaces. Basic services, such as transfer, last number redial, and single key access through the PSTN to a centralized voice messaging system are supported. Local PSTN access and local Emergency Services access is also supported. No local applications or Business Communication Manager features are supported in Local Mode operation.

Analog devices continue to be under the control of the SRG 50 system. It is the intent of Local Mode to provide continued access to the PSTN for critical calls and emergency services.

In Local Mode, since the SRG 50 handles all call processing, calls between two IP phones at the SRG 50 are handled locally as a simple station-to-station call. When an IP Phone initiates a local PSTN call, the

SRG 50 routes the call to a trunk that is connected to the local PSTN. Incoming DID calls are also handled by the SRG 50 and terminated on the appropriate IP Phone.

In the event of a WAN failure, in Local Mode, the IP Phones do not have access to the main office network over the VoIP trunks. If the appropriate alternate routes are configured, calls will be routed to the main office or other branch offices using the available PSTN trunks.

While in Local Mode, the SRG 50 system continues to monitor for a main office CS 1000 heartbeat signal, and once detected, automatically redirects phones on an individual basis back to Normal Mode of operation. If a call is active, the SRG waits until the call is completed before redirecting the phones; calls in progress are not interrupted. This switch-over occurs almost immediately once the SRG determines that an individual phone can be redirected. This reinstates the CS 1000 normal user interface and feature set for the IP Phone user, on a user by user basis.

The SRG 50 system implements the same interface used by the MG 1000B system to interact with the main office CS 1000 system. This allows the main office to identify attached clients and the local PSTN as branch office entities, enabling proper operation of dial plans and E911 access.

In Local Mode, devices that are physically located at the branch office, that are controlled by the local system and receive a basic telephony feature set, provide business continuity for the branch office during the WAN or system failure. The SRG supports a main office heartbeat or reliable UDP signaling which automatically reregisters users once WAN or system failure has recovered.

For information about the features supported in Local Mode, see *SRG 50 Configuration Guide* (NN40140-500) .

Survivability

SRG is specifically designed to provide automatic survivability against WAN failure, main office Call Server failure, main office Signaling Server failure, and Gatekeeper failure.

SRG supports the Geographic Redundancy feature. For further information about Geographic Redundancy, see *System Redundancy Fundamentals* (NN43001-507) .

In the event of a WAN failure, the SRG IP Phones lose communication with the main office. This causes the SRG IP Phones to reset and register with the SRG. The IP Phones then operate in Local Mode, providing basic telephony services delivered by the local SRG system. For further information about services and features supported on the SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

If the main office Call Server fails and call processing services are provided by an Alternate Call Server, the SRG IP Phones reset and reregister with the Alternate Call Server and receive call processing services from it. If no Alternate Call Server is available, the SRG IP Phones go to Local Mode while the SRG attempts to find an Alternate Call Server by way of the NCS.

If the main office Signaling Server fails and an Alternate Signaling Server is available, the SRG IP Phones reset and reregister with the SRG. The SRG will then query the NCS for the Alternate Signaling Server IP address. The SRG will redirect the IP Phone to the Alternate Signaling Server and continue to receive call processing services from the main office Call Server. If no Alternate Signaling Server is available, the SRG IP Phones reset and register with the SRG in Local Mode.

When an IP Phone at the SRG first boots up, the IP Phone attempts to communicate with the SRG. After communication with the SRG is established, the SRG redirects the IP Phone to the main office. When the SRG IP Phone attempts to register with the main office, the SRG first queries the Primary NCS for the main office Virtual Trunk node IP address to redirect the IP Phone. If the Primary NCS is down or unreachable, the SRG queries the Alternate NRS (H.323 Gatekeeper/SIP Redirect Server), if one is specified. If it receives a positive response, the SRG IP Phone is redirected to the specified main office. Otherwise, if neither a Primary or an Alternate NRS (H.323 Gatekeeper/SIP Redirect Server) is available, the SRG IP Phone remains in Local Mode, and receives call processing services from the SRG until communication can be reestablished.

SRG IP Phones in Normal Mode remain registered with the main office if the Primary NCS fails and no Alternate NCS is available. They can call any main office telephone or IP Phones in Normal Mode in other branch offices. However, they cannot call any SRG analog (500/2500-type) telephones or any external numbers through the SRG trunks because an H.323 Gatekeeper/SIP proxy server, which could route call properly in case of an NRS failure, is not available.

Recovery to Normal Mode

After communication is reestablished with the main office call server, all IP Phones at the branch office that are in Local Mode automatically redirect and reregister to the main office and return to Normal Mode operation. IP Phones that were busy at the time communication was reestablished complete the call in Local Mode, and then reregister with the main office after the call is complete.

Local Mode operation

When an SRG IP Phone is in Local Mode, the user has full access to the services configured at the SRG (analog devices or analog or digital trunks) and to other IP Phones registered to the SRG. In Local Mode, the IP Phones can make local calls to other IP Phones and other analog (500/2500-type) telephones at the branch office. They can also be used to make outgoing PSTN calls and receive incoming calls as usual. SRG IP Phones can access the main office IP Phones or other branches by routing through the local PSTN.

Testing the phone in Local Mode

From Normal Mode, the branch user has the option of going to Local Mode manually using the Test Local Mode feature, or when the telephone is power-cycled. The test can be performed by the user at any time and does not require a password. This test is invoked from any IP Phone at the branch office.

Nortel recommends testing Local Mode operation after changing the provisioning for a telephone on the SRG.

To ensure that users do not forget to resume Normal Mode operation, the SRG redirects the telephone to the main office to return the telephone to Normal mode. This occurs if the telephone remains registered to the SRG in Test Local Mode for ten minutes (default setting). Alternatively, the user can press the Quit key on the phone to return to Normal Mode.

For further information about Local Mode functionality for SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Virtual Trunks

In order for endpoints in the CS 1000 network to access endpoints in local mode at the SRG or to access the PSTN at the SRG, Virtual Trunks are used over the LAN/WAN.

Virtual Trunks are software components that provide the trunking features of the Meridian Customer-Defined Network (MCDN) feature set. Access to PSTN digital or analog trunks at the branch office occurs through the MCDN Virtual Trunk.

Virtual Trunks are sometimes referred to as SIP or H.323 Virtual Trunks. In the *SRG 50 Configuration Guide* (NN40140-500) Virtual Trunks are referred to as IP Trunks.

For more information about Virtual Trunks, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

IP Phone calls

When an IP Phone calls another IP Phone, each telephone receives the address of the other to exchange media directly between the telephones. When in Normal Mode, an SRG IP Phone calling a main office IP Phone does not require any trunking to set up the call. However, LAN/WAN bandwidth is used to provide a media path for the call. For more information on Direct IP media path functionality, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

Bandwidth Management

For information about Bandwidth Management, see *Converging the Data Network with VoIP Fundamentals* (NN43001-260) .

Capacity

Each CS 1000 main office can support up to 255 branch offices, which can be made up of any combination SRG and MGC based MG 1000B. SRG 50 Release 2.0 and later supports up to 80 survivable IP users. However, since all IP Phones register with the main office, the governing factor is the maximum number of IP Phones that can be supported at the main office. This means the total number of IP Phones in all offices can be no greater than the capacity of the main office. See one of the following documents to determine the total number of phones your system can support:

- *Communication Server 1000E Planning and Engineering* (NN43041-220)
- *Communication Server 1000M and Meridian 1 Large System Planning and Engineering* (NN43021-220)

Virtual Trunks capacity

The SRG capacity to support a number of simultaneous calls depends on the specific codec type used and the available bandwidth.

If both the intrazone and interzone codes are configured as Best Quality (G.711), the SRG supports up to 24 Virtual Trunks (H.323 or SIP), otherwise, only 15 Virtual Trunks (H.323 or SIP) are supported.

In Normal Mode, the codec selection used is controlled by specific programming of the CS 1000.

In Local Mode, if the WAN has failed, Virtual Trunks between the SRG and CS 1000 cannot be established. However, the SRG will continue to convert calls from IP terminals for communication through the PSTN. Nortel recommends you use G.711 codec.

Branch office dialing plan

Since IP Phone users can be located at a branch office equipped with an SRG, the routing of calls to the local gateway is important (especially when toll charges apply to calls made from the central Call Server that controls the telephone). The administrator can configure digit manipulation through zone attributes for IP Phones to select a main office or branch office that provides PSTN access local to the destination of the call.

Calls from the PSTN to users within the network can be routed with the various ESN numbering plan configurations.

To access local PSTN resources, outgoing calls can be routed using ESN as well as zone parameters that enable digit insertion. The zone parameters force calls made by an SRG user to be routed to the desired local PSTN facilities.

> **ATTENTION**
> Outgoing calls can include local and, optionally, long distance calls.

Nortel recommends that the Branch User ID (BUID) be the same at the branch office as the DN at the main office. A BUID has a maximum of 15 digits. Under the recommended Coordinated Dialing Plan (CDP), the BUID can be an extension (for example, 4567). Under the Uniform Dialing Plan (UDP), the BUID is the user main office DN, the Location Code (LOC), plus the Access Code (for example, 6 343-5555). The main office DN must be an ESN compliant DN. See "ESN Access Codes" (page 24).

The SRG only supports only one dialing plan option at a time. CDP and UDP dialing plan options cannot be configured at the same time in the same system.

For more information about dialing plans and configuration, see "Dialing Plan configuration" (page 39).

ESN Access Codes

ESN data is configured with two Access Codes, called AC1 and AC2. AC1 normally applies to long distance calls, whether placed on or off the customer's private network (for example, dialing 6). AC2 normally applies to local calls (for example, 9). For more information, see *Electronic Switched Network Reference—Signaling and Transmission* (NN43001-280) .

Music on Hold

For SRG users in Normal Mode, the main office provides music to the user if Music on Hold is provisioned. The use of the G.729A/AB codec between the main office and the branch office can impact the music quality.

> **ATTENTION**
> G.723 codec is not supported on SRG 50.

Branch office and SRG 50 terminology

Table 3 "Branch office and SRG 50 terminology" (page 25) lists configuration-related terms and contexts where branch office and SRG 50 terminology differ.

Table 3
Branch office and SRG 50 terminology

Term or context	Branch office	SRG 50
dialing plan	on-net/off-net dialing	Private/Public network dialing
routing	distant steering codes (DSC), Trunk steering codes (TSC), Local steering codes (LSC)	call routing, destination codes, line pool access codes
	Digit manipulation table	dial-out digits (routing)
alternate routing selection	Facility Restriction Level (FRL)	scheduled call routing
Type of number	CDP/UDP/TNDN	CDP/UDP/no equivalent
Numbering Plan ID	ISDN/Telephony (E.164), Private, Telephony (E.163), Telex, (F.69), Data (X.121), National Standard	Private
User ID	BUID	BUID
	bandwidth management zone	Zone ID
Trunks	public exchange	PSTN
	virtual trunk	IP trunk
access codes (SRG 50: destination codes)	7 = system trunk access	7 = not assigned
	8 = Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS)	8 = not assigned 9 = line pool A access code
	9 = public exchange access	
	Network Class of Service (NCOS)	
telephone numbers (internal, not PSTN)	DN	DN

Limitations

The following is a list of limitations for SRG 50 Release 3.0:

- When an IP Phone is in Local Mode, the SRG 50 does not provide all the features as those provided by the CS 1000 main office. In Local Mode, the SRG provides basic features, basic call handling, and basic routing capabilities only.
- When an IP Phone is in Local Mode, the SRG 50 does not support IP Phone Key Expansion Module or Expansion Module for IP Phone 1100 Series.
- You cannot configure the BUID and MOTN using the IP Phone. Configure the BUID and MOTN using SRG Element Manager.
- The SRG and the CS 1000 are configured separately. There is no single management paradigm or application to update both the CS 1000 and the SRG. Use Element Manager to configure the SRG, and use standard configuration tools to configure the CS 1000.
- Virtual Office Login is not supported in Local Mode.
- Language, Volume, and Contrast settings in the SRG are not synchronized with the CS 1000 settings which causes a potential mismatch in settings between Normal Mode and Local Mode.
- Language options available on the CS 1000 may not be available on the SRG.
- For the Alternate Routing for Network Bandwidth Management feature, the SRG does not support an automatic redirection of IP trunk calls through the PSTN when such calls are blocked by the CS 1000 due to bandwidth availability.
- Multiple ESDN is not supported.
- VLAN tagging is not supported. However, VLAN tagging is achieved by using an external router.
- Active Call Failover is not supported.
- SIP trunks are used only for voice packet traffic alone. H.323 trunking is used for main office and Gatekeeper/NRS discovery, polling of WAN link, as well as voice traffic.

Setting up the main office

Contents

This section contains the following topics:

Introduction

This section describes the following information required to configure the main office:

- SRG information required by the main office
- Main office information required by the SRG
- Zone parameters
- IP Phone passwords and parameters
- Branch office IP Phone configuration

For more information on main office configuration, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

SRG information required by the main office

The main office administrator must gather information about the SRG system. The following information is required:

- an inventory of IP Phones that will be installed on the SRG so the administrator knows what type of telephone to assign to each main office terminal record
- information which allows the administrator to create an NCS (H.323 Gatekeeper or SIP Redirect Server) entry for the SRG
- if using advanced routing, such as tandem dialing between systems, local PSTN number for the SRG and the internal SRG routing codes that will allow the main office to connect to the SRG and to tandem over the SRG PSTN lines, is required

Use Table 4 "SRG information required for the main office configuration" (page 28) to record the information before setting up the SRG on the main office server.

Table 4
SRG information required for the main office configuration

SRG parameters	Information about this system
SRG public IP address	
H.323 ID (required for requests to NCS) Each H.323 ID in the node should match SIP endpoint name for this system in pure SIP environment.	
List of types and number of IP Phones Telephone types are hard-coded to the Terminal Numbers (TN) and the main office. Therefore, install the same type of IP Phones to the coordinating record on the SRG.	
PSTN number to dial into the SRG (in local mode)	
Destination codes (steering codes) to route the main office calls to the SRG and out through the SRG PSTN lines	
IP Ports that affect SRG traffic with the main office and have been assigned firewall filters For further information on port configuration, see *Converging the Data Network with VoIP Fundamentals* (NN43001-260) or *SRG 50 Configuration Guide* (NN40140-500) .	

Main office information required by the SRG

The main office administrator must supply numerous main office settings to the SRG installer so that the SRG can be efficiently configured. In addition, the main office administrator needs to supply the following information:

- a list of the terminal record numbers (TN)
- a list of BUID (Prime DN)
- if using advanced routing, such as tandem dialing between systems, main office routing (steering) codes, are required

Use Table 5 "Main office interoperation information" (page 29) to record main office information required by the SRG.

Table 5
Main office interoperation information

Main office components	Information about this system
Main office IP network information:	
Main office call server type	S1000 (default)
Primary network connect server address	
Alternate network connect server	
Network Connect server port	
Trunk/telephony preferred codecs and jitter buffers listed in order of preference	
NRS (H.323 Gatekeeper/SIP Redirect Server) requirements	
Indicate if the SRG needs to manually assign ports with firewall filters.	
Telephony programming:	
DN length, DN (TN) range	
Numbering plan ID	Private (default)
Type of number SRG 50 only supports CDP and UDP dialing plans. Nortel recommends that the SRG use CDP. The SRG supports only one dialing plan option at a time. CDP and UDP dialing plan options cannot be configured at the same time in the same system.	

Table 5
Main office interoperation information (cont'd.)

Main office components	Information about this system
Node ID When the SRG is down the phones use S2 settings to register with the main office.	
Virtual Private Network ID (VPNI)	
Zone ID and dialing string information requirements	
Main office dial-up number (for PSTN calls to the main office in Local Mode)	
Access code to reach the main office PSTN through VoIP trunks	
Zone dialing: • ZDP appended to SRG IP Phone PSTN dialing strings to redirect the call to SRG PSTN • Any steering codes (destination codes) that must be mirrored by SRG programming	
IP Phone configuration:	
MOTN/BUID list, including which type of IP Phone is assigned to each number. Make note of the leading number, as SRG uses this as the DN range for CDP dialing. If the DCP access code is more than one digit, the second digit number must also be used to further define the DN range.	
Current IP Phone firmware version	
Is a VLAN configured on the network?	

Zone parameters

Zone parameters must be configured at both the main office Call Server and the SRG. The main office procedure is similar to an IP Peer Network configuration with the branch office-specific configuration outlined in this chapter.

Zone parameters are defined at the main office in LD 117 and are applied to IP Phones in LD 11.

Use Procedure 1 "Configuring ESN and SRG zones" (page 31) to configure ESN and SRG zones.

Procedure 1
Configuring ESN and SRG zones

> **ATTENTION**
> Before and after an upgrade, perform a data dump (using LD 43 EDD or through Element Manager) on the Call Serve or on the MGC to back up existing data.

Step	Action
1	Configure the Home Location Code (HLOC) and the Virtual Private Network Identifier (VPNI).

Table 6
Configure Customer Data Home Location Code and Virtual Private Network Identifier

Prompt	Response	Description
REQ:	CHG	Changing existing data
TYPE:	NET	ISDN and ESN Networking options
CUST		Customer number
	0-99	Range for Large 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)-16383	Virtual Private Network Identifier for Bandwidth Management feature X = Disables feature 1-16383 = Enables feature <cr> = No Change

2 Configure the zone properties for IP Telephony bandwidth management. Use LD 117 or Element Manager. See *IP Peer Networking Installation and Commissioning (NN43001-313)*.

The branch office zone number and zone bandwidth management parameters at the main office must match the corresponding branch office zone number and zone bandwidth management parameters at the branch office.

> **ATTENTION**
> Zone 0, the default zone, must not be configured as a branch office zone. Network Bandwidth Management does not support zone 0. If zone 0 is configured as an branch office zone, the Bandwidth Management feature is not activated.

3. Define the zone parameters for the branch office. Use LD 117 or Element Manager. See *IP Peer Networking Installation and Commissioning (NN43001-313)*.

Table 7
LD 117 Define zone parameters for the branch office

Command	Description
CHG ZBRN <Zone> <yes|no>	Define a zone as a branch office zone.
CHG ZDST <Zone> <yes|no> <StartMonth> <StartWeek> <StartDay> <StartHour> <EndMonth> <EndWeek> <EndDay> <EndHour>	If the branch office 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> <TimeDifferencefromMainOffice>	Specified in minutes, the time difference between main office and branch office when both are not in DST.
CHG ZDES <Zone> <ZoneDescription	A name to render data display more meaningful.

4. Enable the features for the branch office zone in LD 11.

Table 8
LD 117 Enable features for an SRG zone

Command	Description
ENL ZBR <zone> ALL	Enables features for branch office <zone>.

--End--

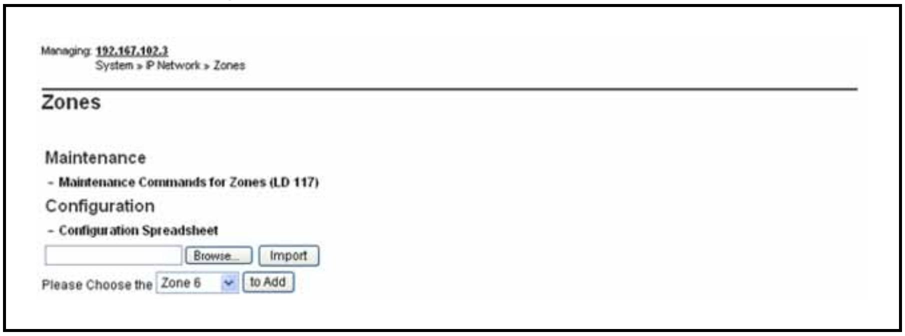
Configuring zone parameters using CS 1000 Element Manager

Use Element Manager to configure the branch office specific zone properties and time difference.

1. Select **IP Network > Zones** in Element Manager navigator.

   The Zones window opens. See Figure 2 "Zone List web page" (page 33). The zone list is the main window used for zone configuration.

Figure 2
Zone List web page



2. Select the zone to be configured and configure the following properties.

   — Basic Property and Bandwidth Management (see Figure 3 "Zone Basic Property and Bandwidth Management web page" (page 33))

   — Time Difference and Daylight Saving Time Property (see Figure 4 "Zone Time Difference and Daylight Saving Time web page" (page 34))

Figure 3
Zone Basic Property and Bandwidth Management web page

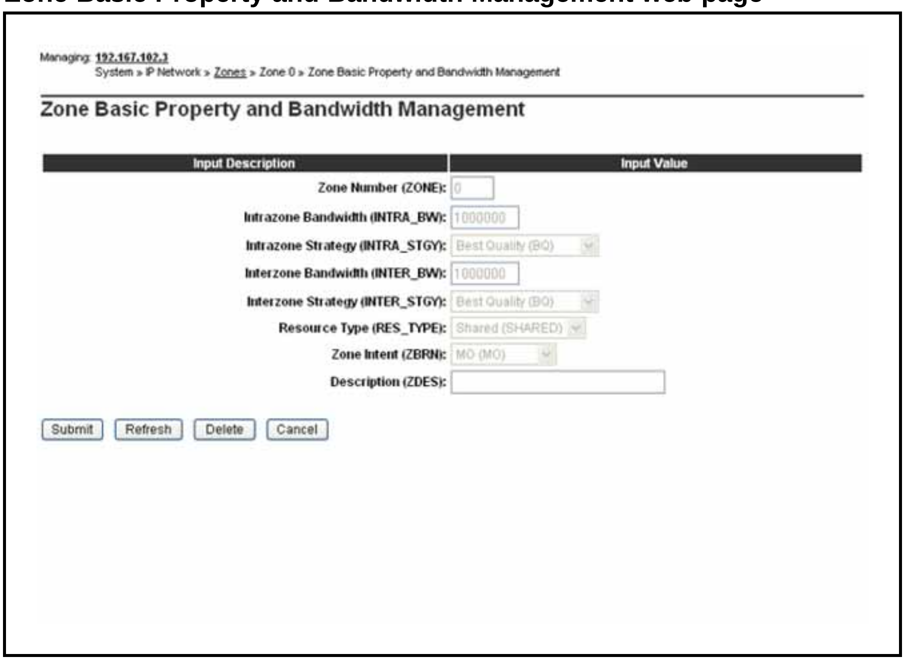
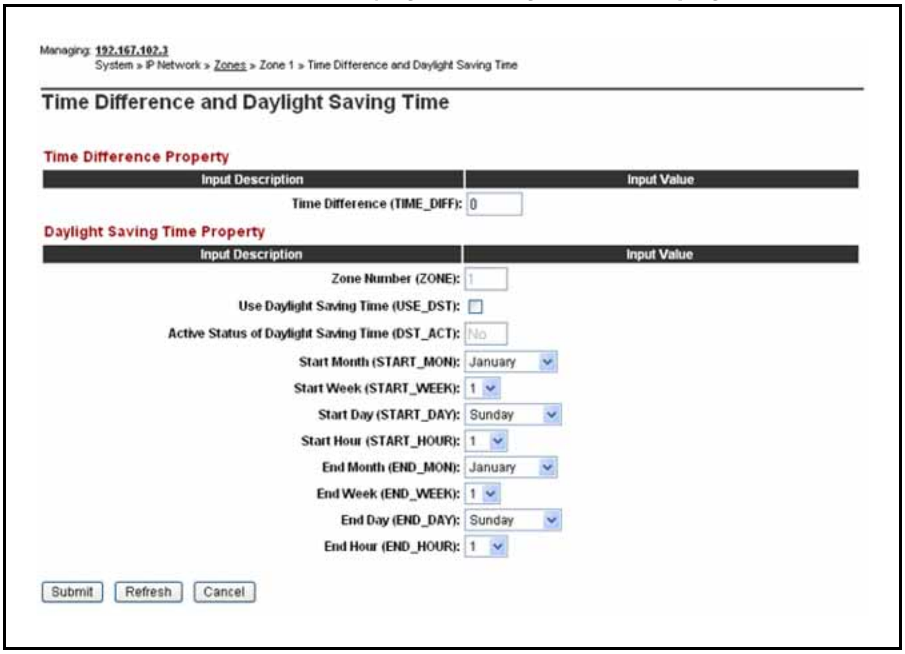


Figure 4
Zone Time Difference and Daylight Saving Time web page



Zone parameters must be configured on the main office and the branch office. For information on configuring zones, see Bandwidth Management.

Branch office IP Phone configuration at the main office

After the branch office zones and passwords are provisioned, provision the branch office IP Phones at the main office. These can be provisioned using Telephony Manager 4.0. See "Branch office IP Phone configuration using Telephony Manager 4.0" (page 34)or LD 11. See Procedure 2 "Configuring branch office IP Phones at the main office using LD 11" (page 35).

> **ATTENTION**
> There is no automatic data synchronization between the main office Call Server and SRG. The technician must provision the telephone on both the Call Server and the SRG.

Branch office IP Phone configuration using Telephony Manager 4.0

At the main office, Telephony Manager 4.0 can be used to configure branch office IP Phones. Use Telephone Pages to configure the telephones to include the following:

- Terminal Type
- TN

- Customer Number
- Branch Office Zone
- Prime DN corresponding to the BUID

See *Telephony Manager 4.0 System Administration* (NN43050-601) for details.

Branch office IP Phone configuration using LD 11

Use Procedure 2 "Configuring branch office IP Phones at the main office using LD 11" (page 35) at the main office to configure branch office IP Phones.

Procedure 2
Configuring branch office IP Phones at the main office using LD 11

Step	Action
1	Configure the branch office zones and dialing plan. See Procedure 1 "Configuring ESN and SRG zones" (page 31).
2	Configure the following telephone data in LD 11: • Terminal type • Customer Number • TN • Zone • Prime DN to correspond to BUID

Table 9
LD 11 Provision Branch User and SCPW at the main office

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	a...a	Terminal type. Type **?** for a list of possible responses.
CUST	xx	Customer number as defined in LD 15.
ZONE	0-255	Zone number to which the IP Phone belongs. The zone prompt applies only when the TYPE is 2001P2, 2002P2, 2004P2, 2050PC, 2007, 1110, 1120, 1140, 2210, 2211, 2212, 1210, 1220, 1230 Zone number is not checked against LD 117.

Table 9
LD 11 Provision Branch User and SCPW at the main office (cont'd.)

Prompt	Response	Description
...	...	...
SCPW	xxxx	Station Control Password Must equal Station Control Password Length (SCPL) as defined in LD 15. Not prompted if SCPL = 0. Precede with X to delete.

--End--

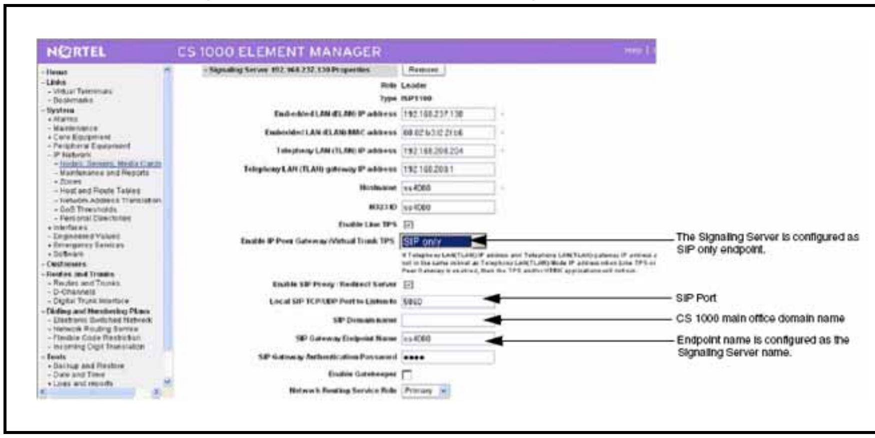
SIP IP Trunks configuration at the main office

In order for the SRG 50 to act as a SIP endpoint and to use the SIP Trunks for call signaling with the CS 1000, you must configure SIP Trunks between the SRG 50 branch office and the main office.

Configuring SIP IP Trunks

Step	Action
1	From the Element Manager navigator, click **IP Network** > **Nodes: Servers, Media Cards**. The Node Configuration window appears.
2	Click the **Edit** button associated with the node to be updated.
3	Click the plus (+) sign beside Signaling Server Properties.
4	From the Enable IP Peer Gateway (Virtual Trunks TPS) list, select **SIP only**.
5	Enter the CS 1000 domain name in the SIP Domain Name field.
6	Enter the SIP Port number in the **Local SIP TCP UDP Port to Listen to** field.
7	Enter the Signaling Server name in the **SIP Gateway Endpoint Name** field. See Figure 5 "SIP Trunk configuration in Element Manager" (page 37).

Figure 5
SIP Trunk configuration in Element Manager



8 Click **Save and Transfer**.

The Save and Transfer window appears.

9 Click **OK**.

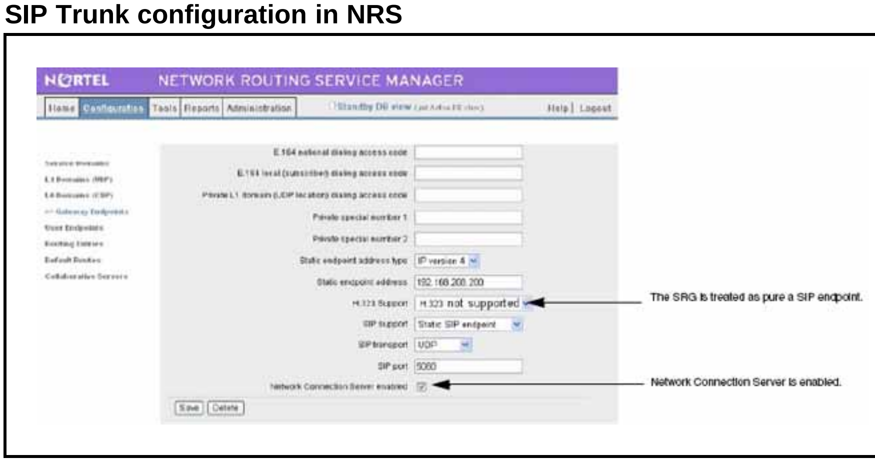
10 Log on to Network Routing Service (NRS) Manager.

11 Select the **Configuration** tab.

12 From the H.323 Support list, select **H.323 not supported**.

13 Select the **Network Connection Server enabled** check box. See Figure 6 "SIP Trunk configuration in NRS" (page 37).

Figure 6
SIP Trunk configuration in NRS



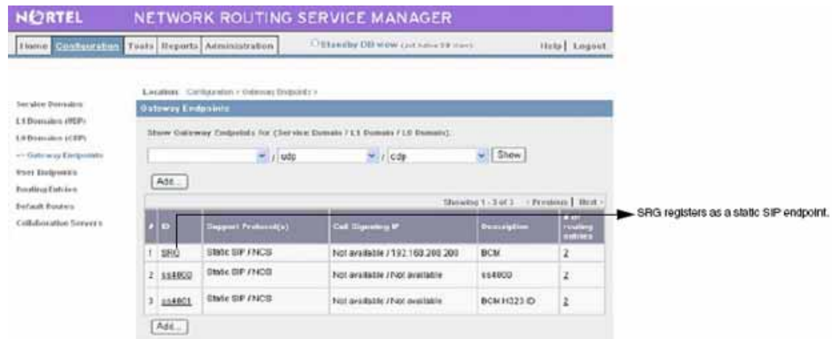
14 Select **Save**.

15 Select **Configuration** > **Gateway Endpoints**.

The Gateways Endpoints window appears.

The SRG registers as a static SIP endpoint. See Figure 7 "Gateways Endpoints window in NRS" (page 38).

Figure 7
Gateways Endpoints window in NRS



--End--

Dialing Plan configuration

Contents

This section contains the following topics:

Overview

This section provides an overview of dialing plan programming on the SRG and the main office.

When a number is dialed, the Call Server determines whether the called number is internal or external to the branch office. If internal or off-net, the system terminates the call on the appropriate terminal. If external or on-net, the system routes the call using one of the supported dialing plans.

On-net dialing plan

The SRG only supports only one dialing plan option at a time. CDP and UDP dialing plan options cannot be configured at the same time in the same system.

The SRG supports the following dialing plans:

- Coordinated Dialing Plan (CDP) – BUID is the same as the Directory Number (DN)
- Uniform Dialing Plan (UDP) – Location code is added to the DN for the BUID

> **ATTENTION**
> Nortel recommends that the SRG use CDP.

CDP Terminal Numbers (TN) can be activated on the other systems if the user moves and wants to retain their phone number. SRG does not support Transferable Directory Numbers (TNDN) due to differences in dialing plans and the small range of DN available on the SRG.

For specific examples for CDP and UDP dialing plans, see "Dialing plan examples" (page 43).

Once the call is sent over the IP network, the call is routed to the SRG, which uses the NRS (H.323 Gatekeeper/SIP Redirect Server) to route the call. The NRS (H.323 Gatekeeper/SIP Redirect Server) translates the address form a telephone number to an IP address, and authorizes the call.

Specific dialing plan configuration is required for IP Phones to properly select a main office or a branch office that provides access to the PSTN for the originating IP Phone. A common configuration might be:

- SRG users select the SRG PSTN for local calls.
- Main office users select the main office PSTN for local calls.
- All users select either the main office or SRG PSTN for long-distance calls to minimize toll charges.
- calls configured to minimize toll charges.

However, this configuration represents only one way that the dialing plan could be configured. PSTN calls can be routed according to the point of origin (main office or branch office) and/or the desired destination, and can select trunks at the main office, branch office, or other branch offices as required. Therefore, the user can route calls to gateways that minimize long-distance costs, minimize bandwidth usage, or meet other criteria.

Nortel recommends that customers use Coordinated Dialing Plan (CDP) between the main office and its branch offices since it enables all users, at the main office or the branch office, to call each other using just an extension number. CDP enables consistent dialing between the main office and SRG IP Phones and devices.

For more information, see *Dialing Plans Reference* (NN43001-283) .

Off-net dialing plan

When dialing to the PSTN, the Call Server determines that the call destination is off-net by analyzing the digits that must be preconfigured at major Call Servers in the network.

If routed over a Virtual Trunk, a request is sent to the NRS to determine the location of public E.164 numbers. The NRS is configured with a list of potential alternate routes that can be used to reach a certain dialed number. Each route is configured with a unique route cost to determine the least-cost route.

The NRS replies with the address information for E.164 numbers. It also provides a list of alternative SIP or H.323 endpoints, sorted by cost. If a terminating endpoint resource is busy when a call attempt is made, the originating endpoint tries the next alternative. If no alternative is available over the IP network, the originating endpoint steps to the next entry on its route list, which could be a TIE or PSTN alternate route.

Dialing Plan Overview

Depending upon the type of dialing plan used in the network (Coordinated Dialing Plan [CDP], or Uniform Dialing Plan [UDP] or a combination of both) the general idea is to have all calls that are terminating at a branch office first dial a number that will get routed to the main office associated with that branch office. The main office recognizes this number as belonging to the branch office and appends a tandem prefix to this number using Digit Manipulation Index (DMI). The main office then routes the call to the branch office while accounting for the additional bandwidth used.

See Figure 8 "A call between two branch offices tandems through the main office" (page 42) for an example of a tandem call.

Figure 8
A call between two branch offices tandems through the main office

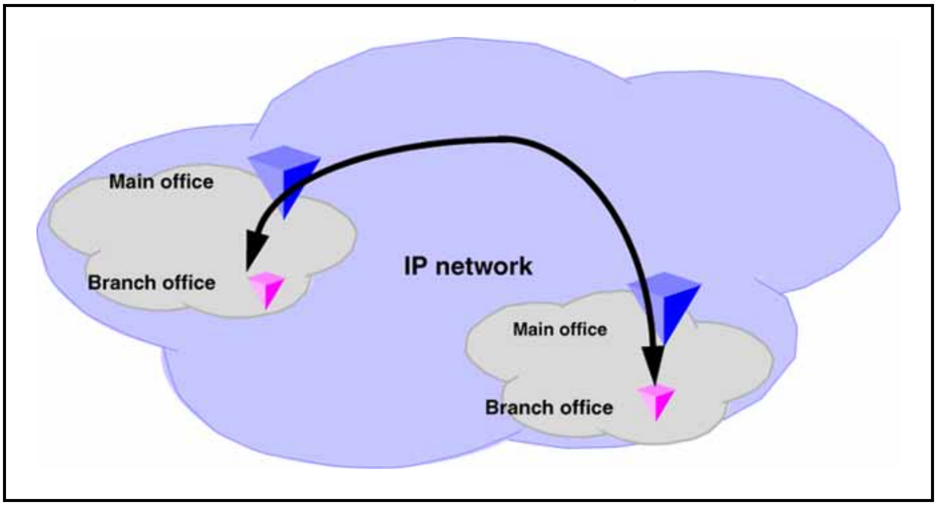
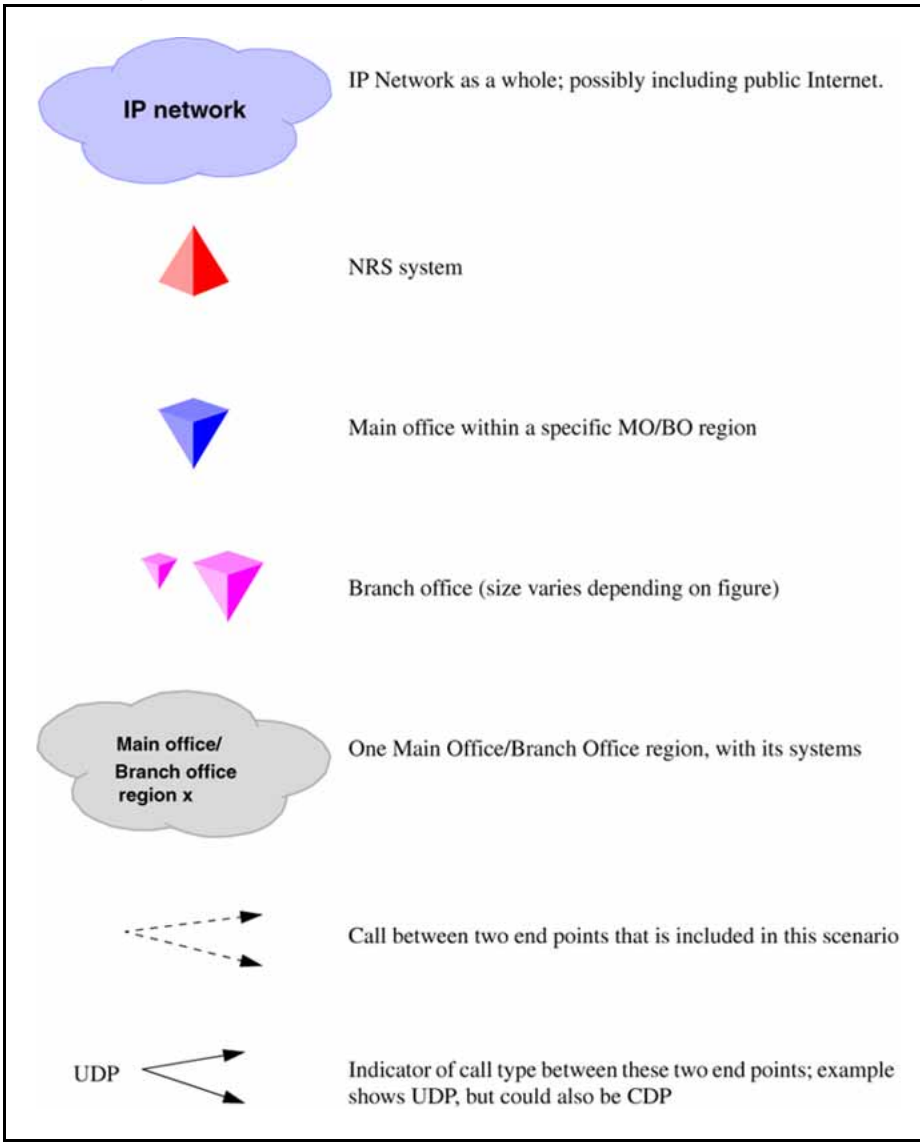


Figure 9 "General legend" (page 43) shows a general legend for the figures in the following section.

Figure 9
General legend



Dialing plan examples

This section describes the following dialing plans:

- Coordinated Dialing Plan (CDP)
- Uniform Dialing Plan (UDP)

Coordinated Dialing Plan

The following section provides three options for creating a CDP dialing configuration.

Overview

Dialing plans between the SRG and the main office need to be coordinated to ensure seamless dialing between the systems. The option you choose will determine how the user dials the other system or the SRG IP telephones.

- Option 1: DN ranges in the main office and SRG are unique, and DNs for SRG IP Phones are the same in both Normal and Local mode. This is the recommended configuration to support seamless dialing on both systems. See “Option 1” (page 49).
- Option 2: DN ranges in the main office and SRG overlap, and DNs for SRG IP Phones are the same in both Normal and Local mode. See “Option 2” (page 54).
- Option 3: DN of SRG IP Phones and DN in the main office overlap in Normal Mode, but are unique in Local Mode. See “Option 3” (page 60).

Call scenarios

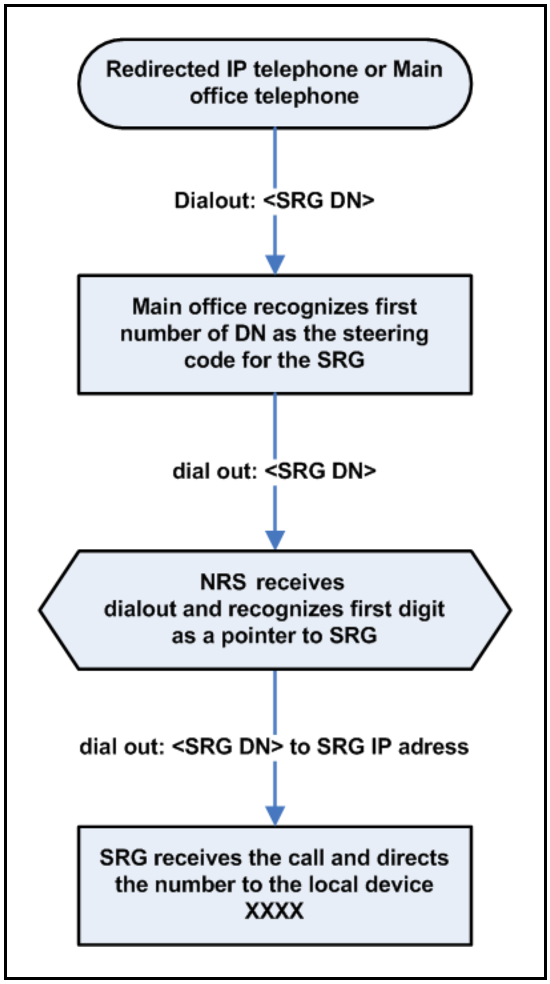
Call scenarios fall into the following categories:

- Common call scenarios occur in all CDP calls, regardless of which option is used.
- Unique call scenarios occur only within certain CDP options.

This section describes the common call scenarios. The unique call scenarios are described with the configuration of the corresponding option, starting with “Option 1” (page 49).

Normal Mode: Main office telephone calls an analog phone at the SRG The call is routed through the NRS and handled by the SRG. Figure 10 "Normal Mode: Main office telephone calls an analog phone at the SRG" (page 45) shows how the call proceeds.

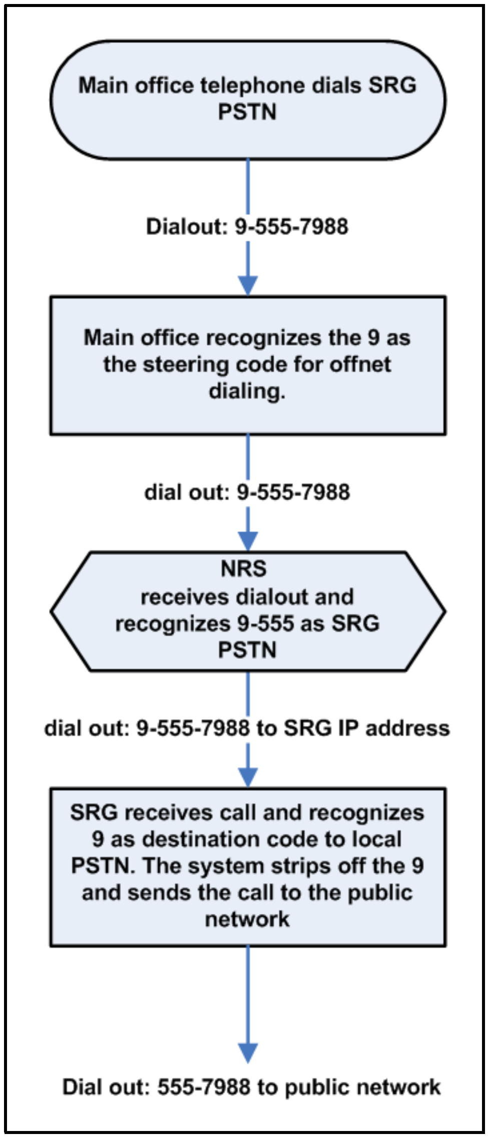
Figure 10
Normal Mode: Main office telephone calls an analog phone at the SRG



Normal Mode: Main office telephone calls a branch IP Phone The call is recognized as a main office number, and the call is directed to the SRG IP telephone using internal routing at the main office.

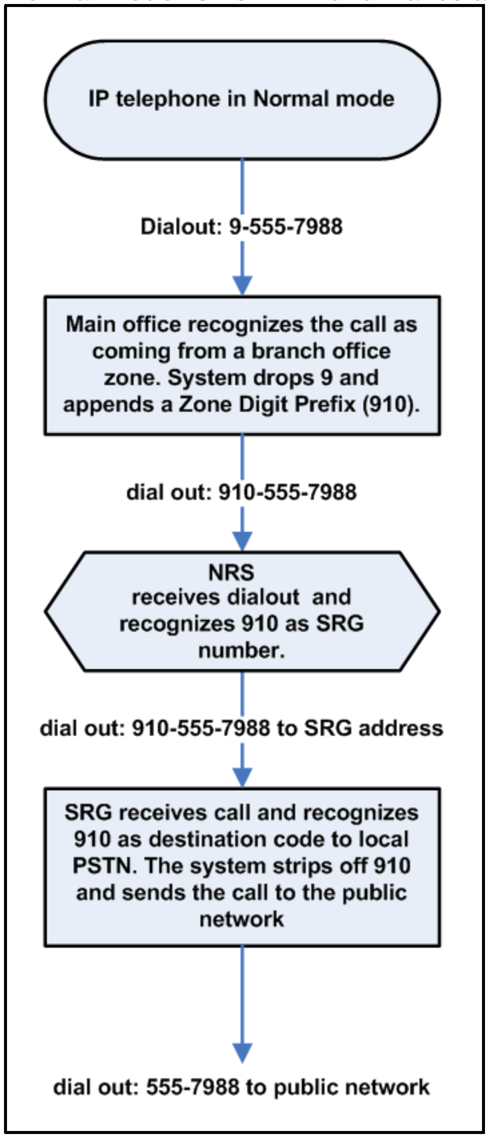
Normal Mode: Main office telephone makes a call over the PSTN through the SRG Routing is configured so the destination code of the PSTN through the SRG is at the start of the dialing string. Figure 11 "Normal Mode: Main office telephone makes a call over the PSTN through the SRG" (page 46) shows how the call proceeds.

Figure 11
Normal Mode: Main office telephone makes a call over the PSTN through the SRG



Normal Mode: SRG IP Phone makes a call over the PSTN Zone management at the main office recognizes that an SRG IP Phone in Normal Mode is dialing the PSTN. Figure 12 "Normal Mode: SRG IP Phone makes a call over the PSTN" (page 47) shows how the call proceeds.

Figure 12
Normal Mode: SRG IP Phone makes a call over the PSTN



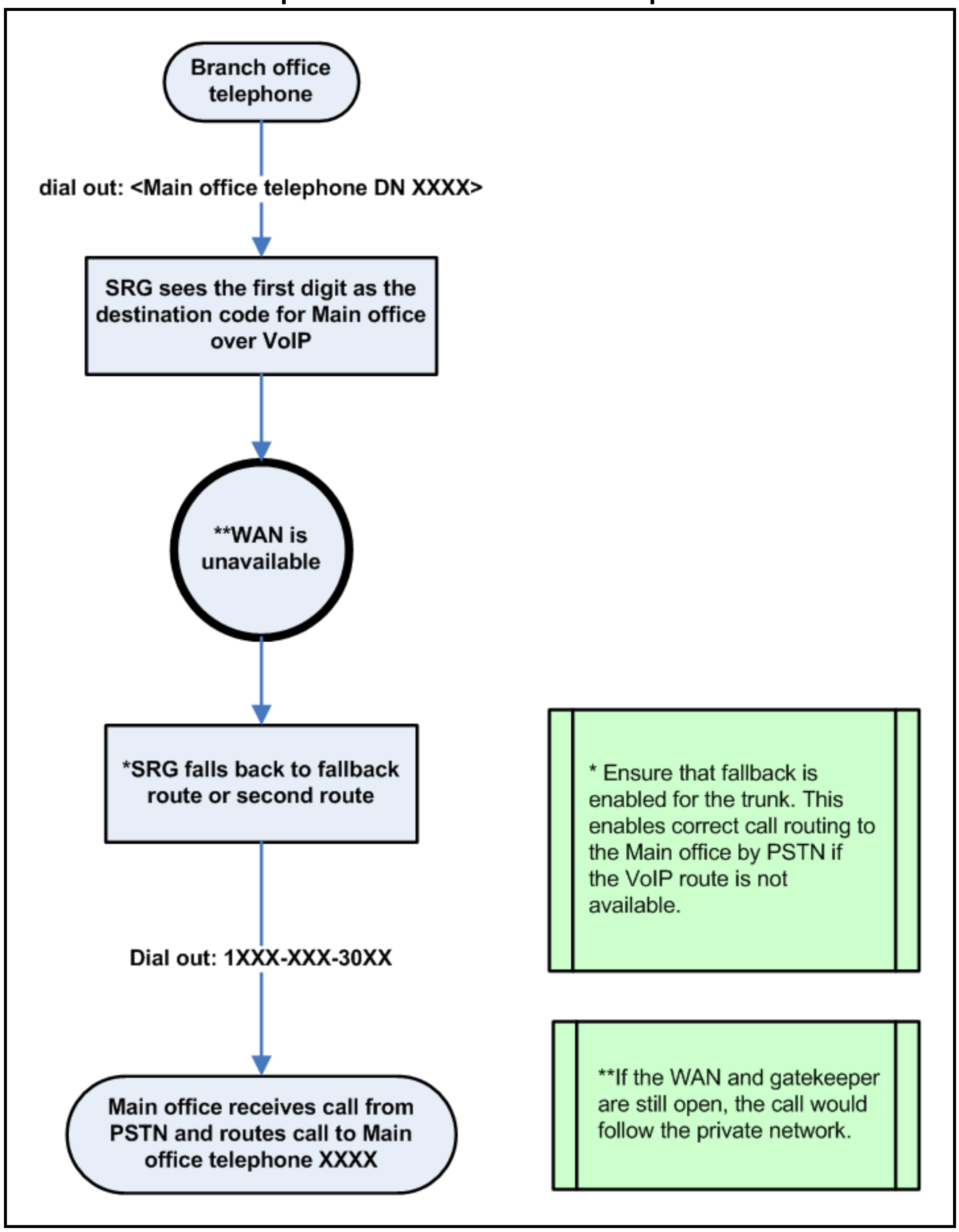
A telephone registered to the SRG calls another telephone registered to the SRG The SRG routes the call internally.

Local Mode: SRG telephone calls an SRG IP Phone The call is handled by the SRG and is sent directly to the SRG IP Phone.

Local Mode: SRG telephone calls a main office telephone In this case, the WAN or NRS is not accessible. Figure 13 "Local Mode: SRG telephone calls a main office telephone" (page 48) shows how the call proceeds.

The user must have configured the fallback route appropriately. See the *SRG 50 Configuration Guide* (NN40140-500) for further information.

Figure 13
Local Mode: SRG telephone calls a main office telephone



Local Mode: Main office telephone calls an SRG IP Phone The call is treated according to main office redirection configuration, such as forwarding to voice mail or continuous ringback.

Option 1

DN ranges in the main office and SRG are unique; DNs for SRG IP Phone are the same in Normal and Local Mode

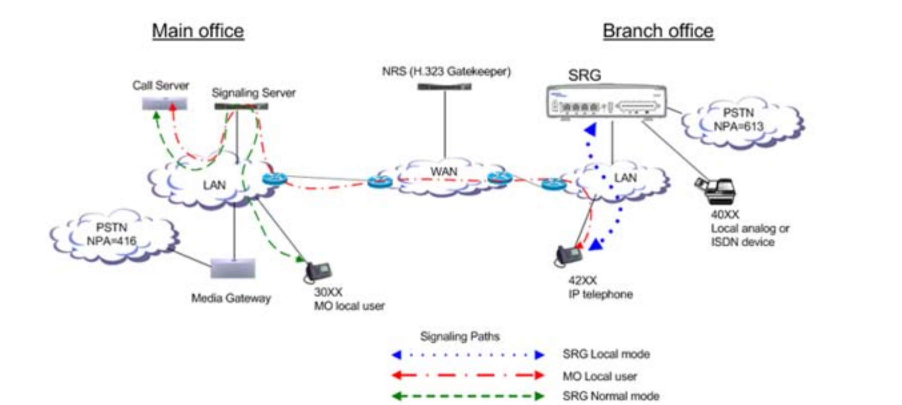
This is the recommended CDP configuration to offer seamless dialing.

In this configuration, the user dials the same DN for SRG IP Phones in either Normal or Local Mode. The DN for SRG IP Phones are configured to be the same on both the SRG and main office. This allows seamless dialing from both the SRG and main office. However, in this configuration, the DN range for telephones registered at the SRG is unique from the DN range for telephones registered at the main office.

The advantage of this configuration is that the system manages the routing for the SRG IP Phones, so users in the SRG and main office do not have to be aware of whether the SRG is in Normal Mode.

See Figure 14 "CDP Option 1" (page 49).

Figure 14
CDP Option 1



MO Call Server Configuration	Gatekeeper Configuration	SRG Configuration
DN: 30XX (zone 10)	CDP Domain "MO_BO_CDP"	DN: 42XX (BUID 42XX)
42XX (zone 15)(SRG)	30XX "MO"	40XX (local service)
CDP: yes	42XX "MO"	CDP: yes
DSC: 40XX	40XX "BO"	VoIP trunk access 3
ZDP (zone 15) 901	901 "BO"	*Dest. Code for VoIP: 30/34

Call scenarios Common call scenarios for this CDP option are listed in "Call scenarios" (page 44). The following additional call scenarios are unique to this CDP option:

- An SRG analog telephone registered to the SRG calls a telephone registered at the main office that can also be an SRG IP Phone in Normal Mode.

Figure 15 "Calls to an SRG IP Phone and a main office IP Phone registered to the main office " (page 51) shows the WAN is up. An SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office (Normal Mode).

Figure 16 "Calls to an SRG analog phone, SRG IP Phone, and a main office IP Phone " (page 52) shows the WAN is down. An SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the SRG (Local Mode).

Figure 15
Calls to an SRG IP Phone and a main office IP Phone registered to the main office

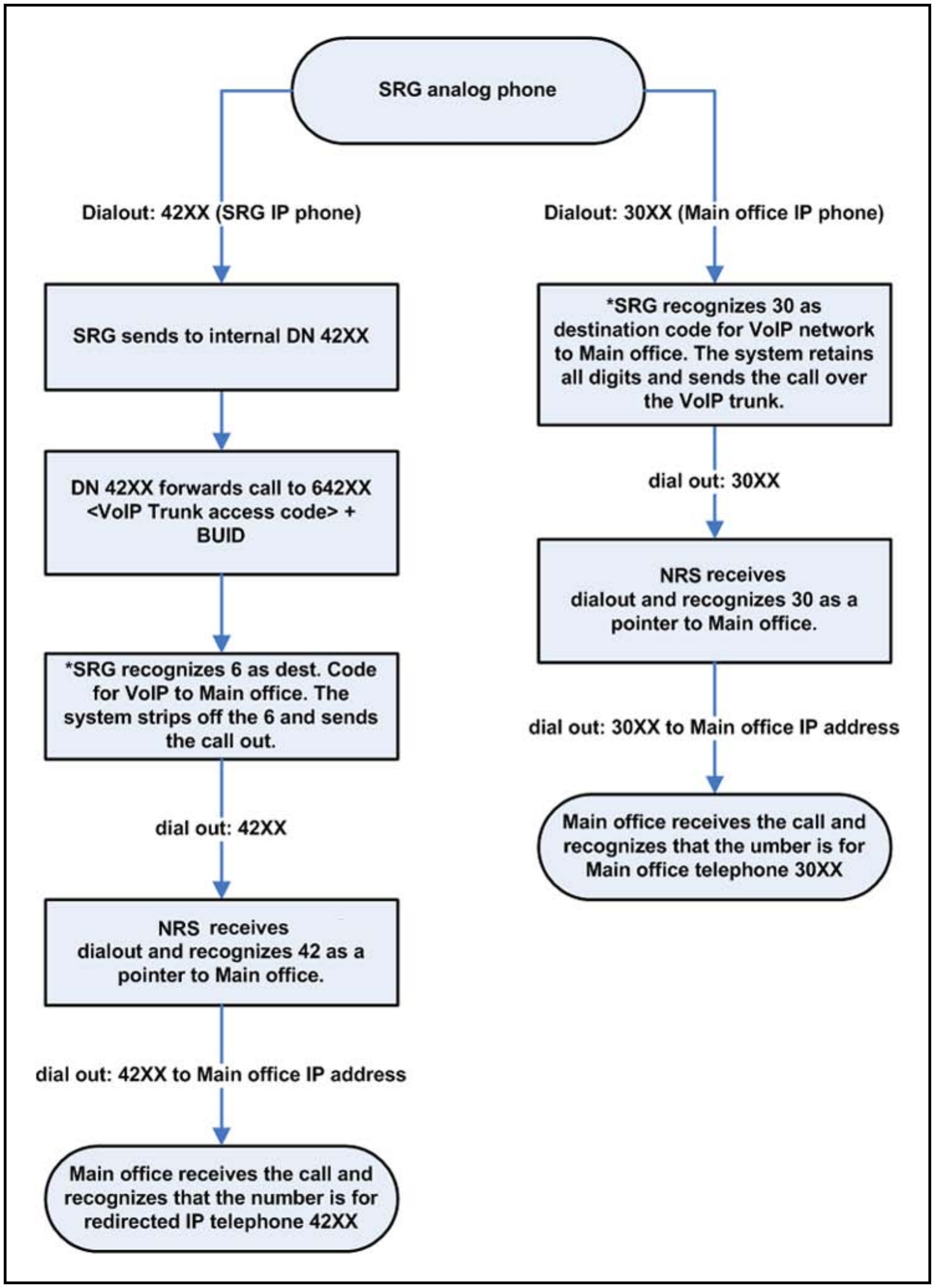
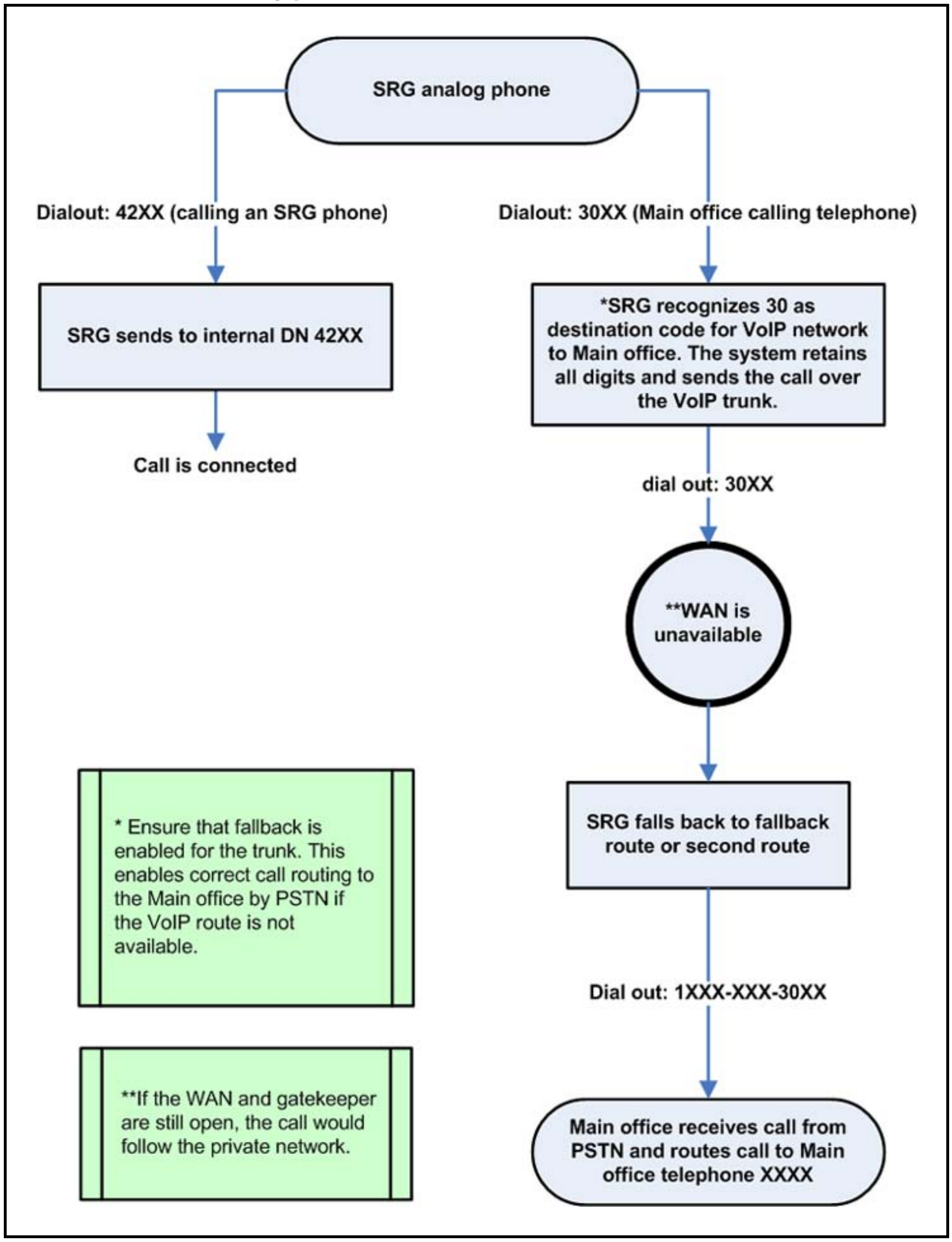


Figure 16
Calls to an SRG analog phone, SRG IP Phone, and a main office IP Phone



Configuration To configure the main office:

- Configure the ESN Control Block for CDP in LD 86.

```
>LD 86
REQ NEW
CUST 0
FEAT ESN
CDP YES
MXSC 50
NCDP 4
DLTN YES
```

- Configure the CDP Distant Steering Code (DSC) in LD 87.

```
> LD 87
REQ NEW
CUST 0
FEAT CDP
TYPE DSC
DSC 50
FLEN 4
RLI 12
```

To configure the NRS (H.323 Gatekeeper/SIP Redirect Server):

- Create CDP Domain: MO_BO_CDP.
- Create H.323/SIP endpoints: MO, BO.
- Create Numbering Plan entries in CDP Domain:
  - — Add 40 for endpoint BO.
  - — Add 30 for endpoint MO.
  - — Add 42 for endpoint MO.

For information about configuring H.323/SIP Redirect Server, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

To configure the SRG:

- Configure DN and BUID as the same number on each of the redirected IP Phones. For example, DN/BUID = 42XX.
- Set the main office VoIP Trunk Access code to 3. For example, main office VoIP trunk access code = 3.
- Set the destination code for the VoIP trunk to 30 (retain all digits) or 34 (remove first digit). For example, BUID dialout = 342XX.

  The VoIP route destination codes 30 (no digits dropped) and 34 (1 digit dropped) route any call that starts with 30 or 34 out of the system over the VoIP trunk to the main office.

The main office access code length is still 0.

- Assign the telephones registered to the SRG (IP Phones or analog [500/2500-type]) telephones to a different range, such as 40XX. See the NRS configuration above.

  The users in both the main office and the SRG dial only the DN for all telephones in the main office and the SRG in both Normal Mode and Local Mode.

For more information on configuring the main office and NRS, see *Branch Office Installation and Commissioning* (NN43001-314) and *IP Peer Networking Installation and Commissioning* (NN43001-313) . For more information on configuring the SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Option 2

DN ranges in the main office and SRG overlap; DNs for SRG IP Phones are the same in Normal and Local Mode

In this configuration, the SRG DN overlap with the main office DN. However, since SRG does not support Vacant Number Routing (VNR), a user registered to the SRG must dial a destination code before the main office DN to call a main office telephone.

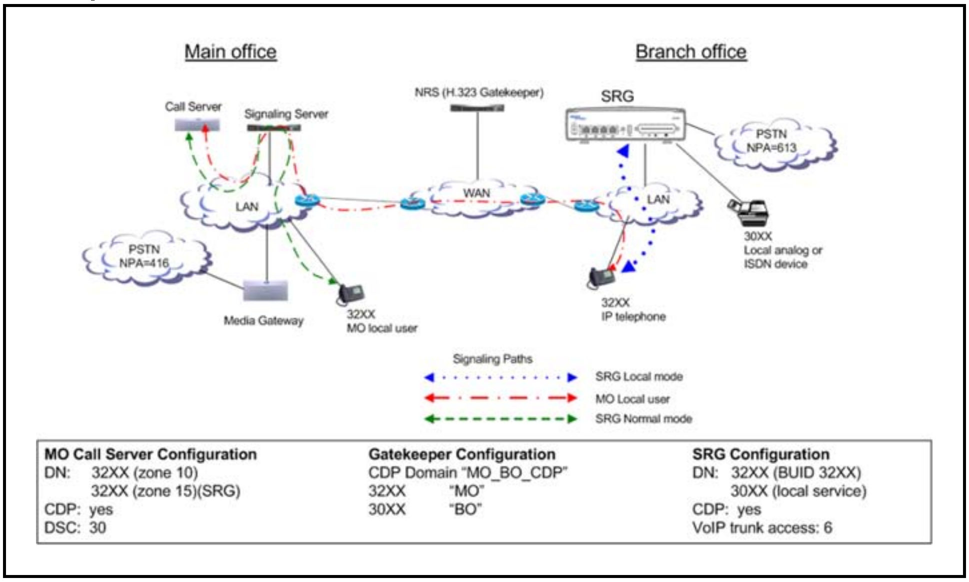
To call an SRG IP Phone in either Normal or Local Mode, SRG and main office users need to dial only the DN for the SRG IP Phone. SRG IP Phone calls are forwarded with the main office Private Network ID/destination code appended to the BUID, which allows the call to flow to the VoIP trunks for the main office.

This configuration is not a true CDP dialing plan. A destination code is added by the system to properly direct the SRG IP Phone calls, since the start digits of the DN are not unique for SRG and main office users. Users dialing a telephone registered at the main office must dial a destination code before the main office DN. This plan allows all systems on the network to appear to be available within a range of numbers.

Since the SRG DN range is limited to about 200 DN, this configuration only works if SRG dialing to the main office is limited to the redirected IP Phones and to a small number of main office telephones, such as to a central attendant and voice mail lines.

See Figure 17 "CDP Option 2" (page 55) shows this CDP option.

Figure 17
CDP Option 2

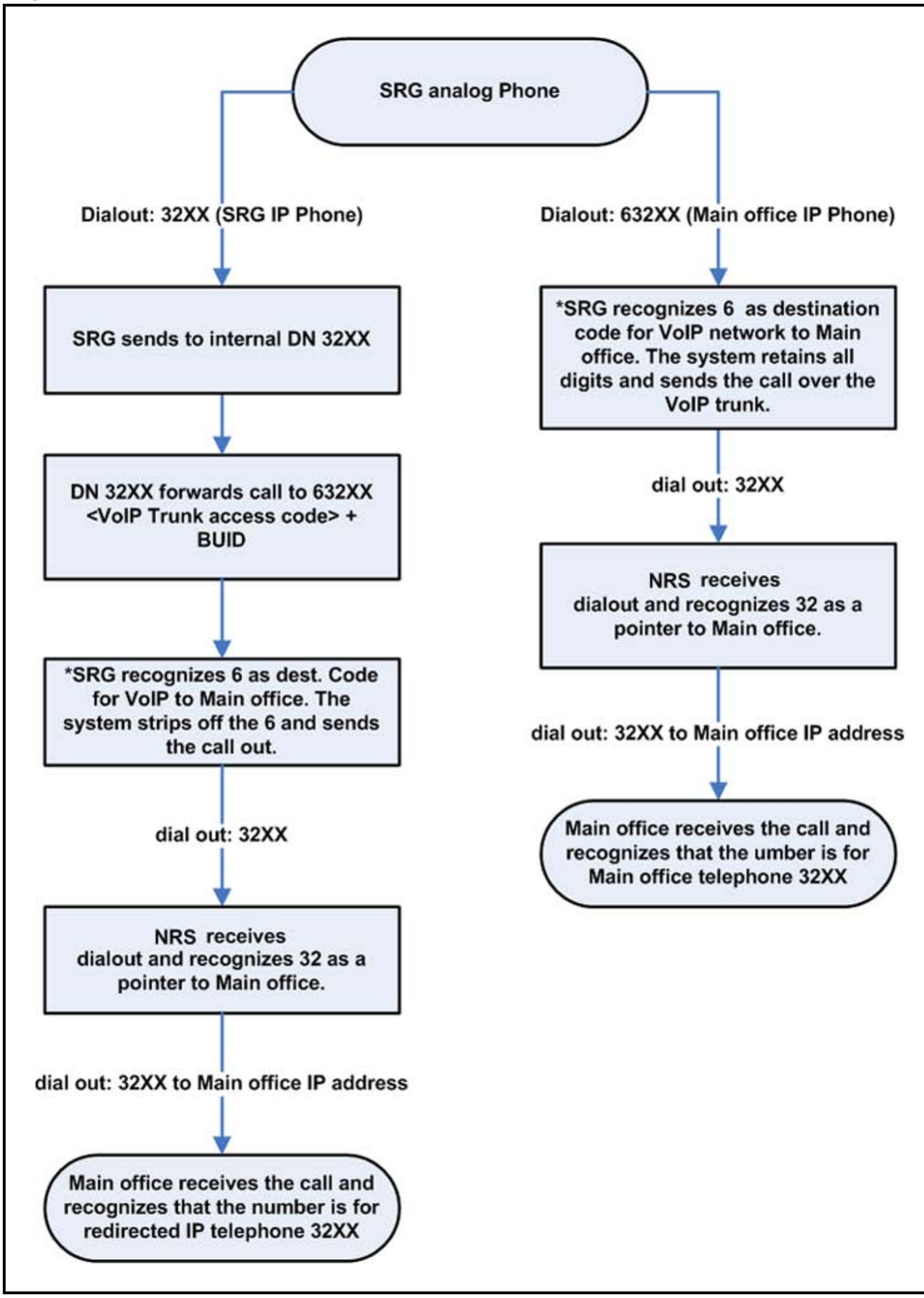


Call scenarios Common call scenarios for this CDP option are listed in “Call scenarios” (page 44). The following additional call scenarios are unique to this CDP option:

- Normal Mode: An SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office.

  The WAN is up. SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main Office (Normal Mode). SeeFigure 18 "SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office" (page 56).

Figure 18
SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office



- Local Mode: SRG IP Phones are registered to the SRG.

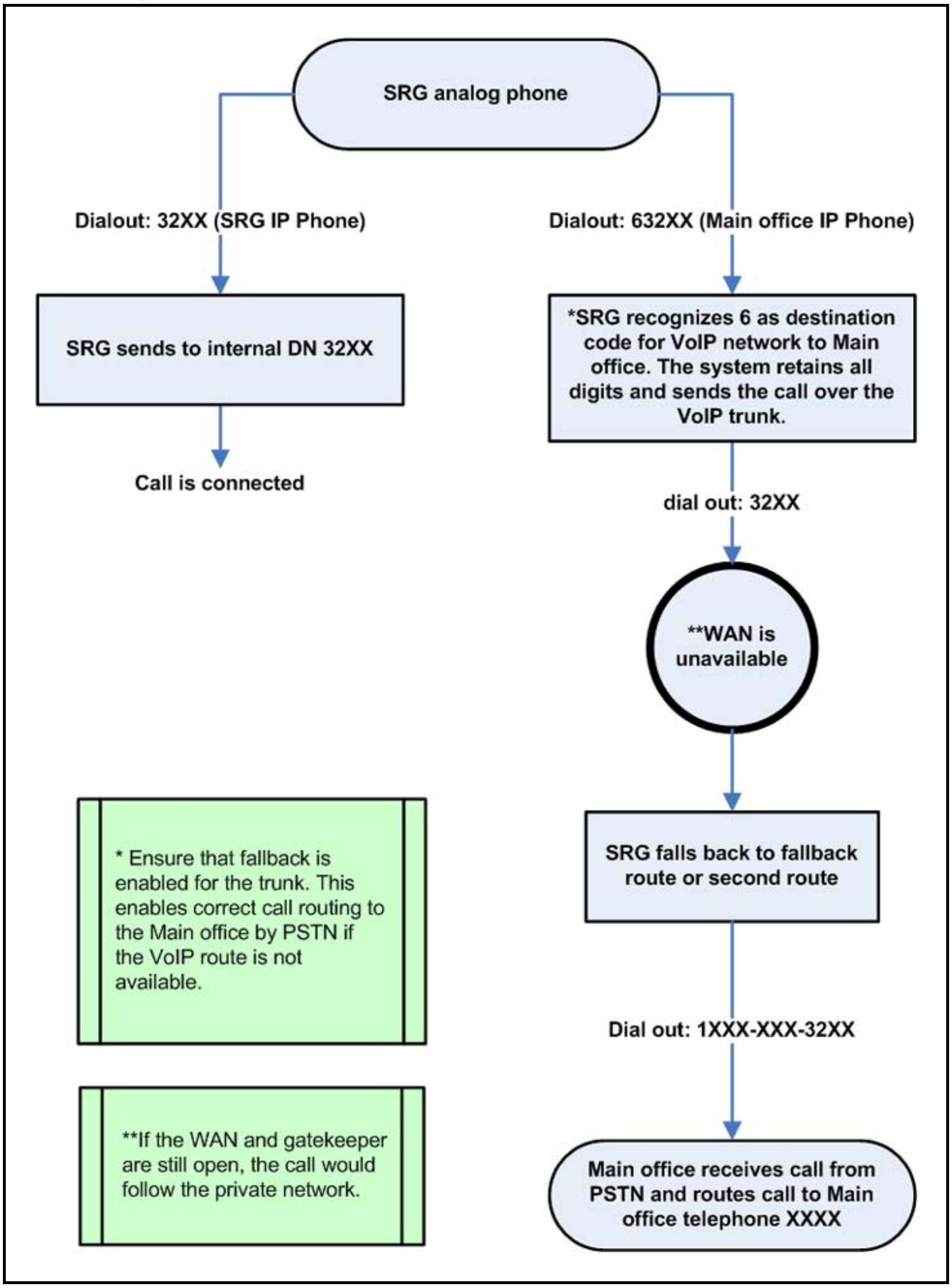
  In this scenario, the WAN and the NCS are working. However, the SRG IP Phones are redirected to the SRG and are in Local Mode (Call Forward All Calls is inactive). The following occur:

— Telephones registered at the SRG dial local DNs (see the common call scenarios given in “Call scenarios” (page 44)).

— SRG calls to the main office use VoIP routing. The WAN is down. SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the SRG (Local Mode) See Figure 19 "SRG analog phone calls an SRG IP Phone and a main office IP Phone " (page 58).

— Main office calls to SRG IP Phones in Local Mode cannot complete because the NRS cannot resolve the numbering.

Figure 19
SRG analog phone calls an SRG IP Phone and a main office IP Phone



Configuration To configure the main office:

- Configure the ESN Control Block for CDP in LD 86.

```
> LD 86
REQ NEW
CUST 0
FEAT ESN
CDP YES
MXSC 50
NCDP 4
DLTN YES
```

- Configure the CDP Distant Steering Code (DSC) in LD 87.

```
> LD 87
REQ NEW
CUST 0
FEAT CDP
TYPE DSC
DSC 50
FLEN 4
RLI 12
```

To configure the NRS (H.323 Gatekeeper/SIP Redirect Server):

- Create CDP Domain: MO_BO_CDP.
- Create H.323 and SIP endpoints: MO, BO.
- Create Numbering Plan entries in CDP Domain:
  — Add 30 for endpoint BO.
  — Add 32 for endpoint MO.

For information about configuring H.323 Gatekeeper/SIP Redirect Server, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

To configure the SRG:

- Configure DN and BUID as the same number on each of the redirected IP Phones. For example, DN/BUID = 32XX.
- Set the main office VoIP Trunk Access code to 6. For example, main office VoIP trunk access code = 6.
- Set the destination code for the VoIP trunk to 6, the same value as the access code. For example, BUID dialout = 632XX.

  The main office access code length is still 0.
- Assign the telephones registered to the SRG (IP Phones or analog [500/2500-type] telephones) to a different range, such as 30XX, than the telephones registered to the main office.

  SRG users must dial the destination code before the DN when making a call to a telephone in the main office, whether they are in Normal or Local Mode. When calling another IP Phone in the SRG, SRG users dial only the DN, whether they are in Normal or Local Mode. The main office uses VNR to route SRG DN to the SRG in both Normal and Local Mode.

For more information on configuring the main office and NRS, see *Branch Office Installation and Commissioning* (NN43001-314) and *IP Peer Networking Installation and Commissioning* (NN43001-313) . For more information on configuring the SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Option 3
DNs of SRG IP Phones and DNs in the main office overlap in Normal Mode, but are unique in Local Mode

In this CDP configuration, each node on the network has unique leading digits that is included in the DN range. The unique leading digits indicate the private network code for the system.

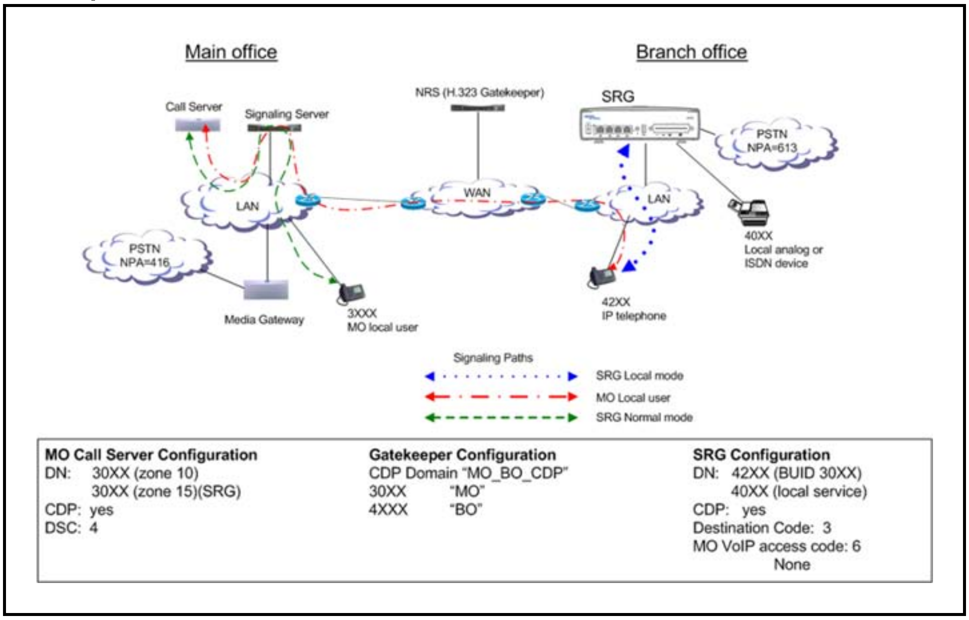
This configuration allows seamless dialing for users registered at the SRG, but main office users must dial a different DN to call SRG IP Phones in Normal and Local mode. Therefore, SRG IP Phones have DNs and BUIDs that do not match.

In Figure 20 "CDP Option 3 " (page 61), the SRG IP Phones have a DN starting with 4 on the SRG to accommodate the SRG Private Network Code. On the main office, the SRG IP Phones are given a DN (BUID) starting with 3, the main office Private Network Code. The NRS is programmed to recognize that 3X numbers go to the main office and that 4X numbers go to the SRG.

In Normal mode, when a call is directed into the SRG, or from a telephone registered at the SRG, to the SRG IP Phone in Normal mode, the SRG system translates the SRG IP telephone DN (4XXX) to the main office BUID (3XXX) so that the call can route correctly through the main office VoIP trunk. Users registered at the main office dial the main office DN (3XXX) for the SRG IP Phone.

In Local mode, the users registered to the SRG still dial the SRG IP Phone DN (4XXX). The main office users can not call the SRG IP Phone by dialing the main office DN for the telephone (3XXX) because the NRS cannot route the call to the SRG. If the main office user dials the SRG IP Phone DN (4XXX), the call goes through.

Figure 20
CDP Option 3



Call scenarios Common call scenarios for this CDP option are listed in "Call scenarios" (page 44). The following additional call scenarios are unique to this CDP option:

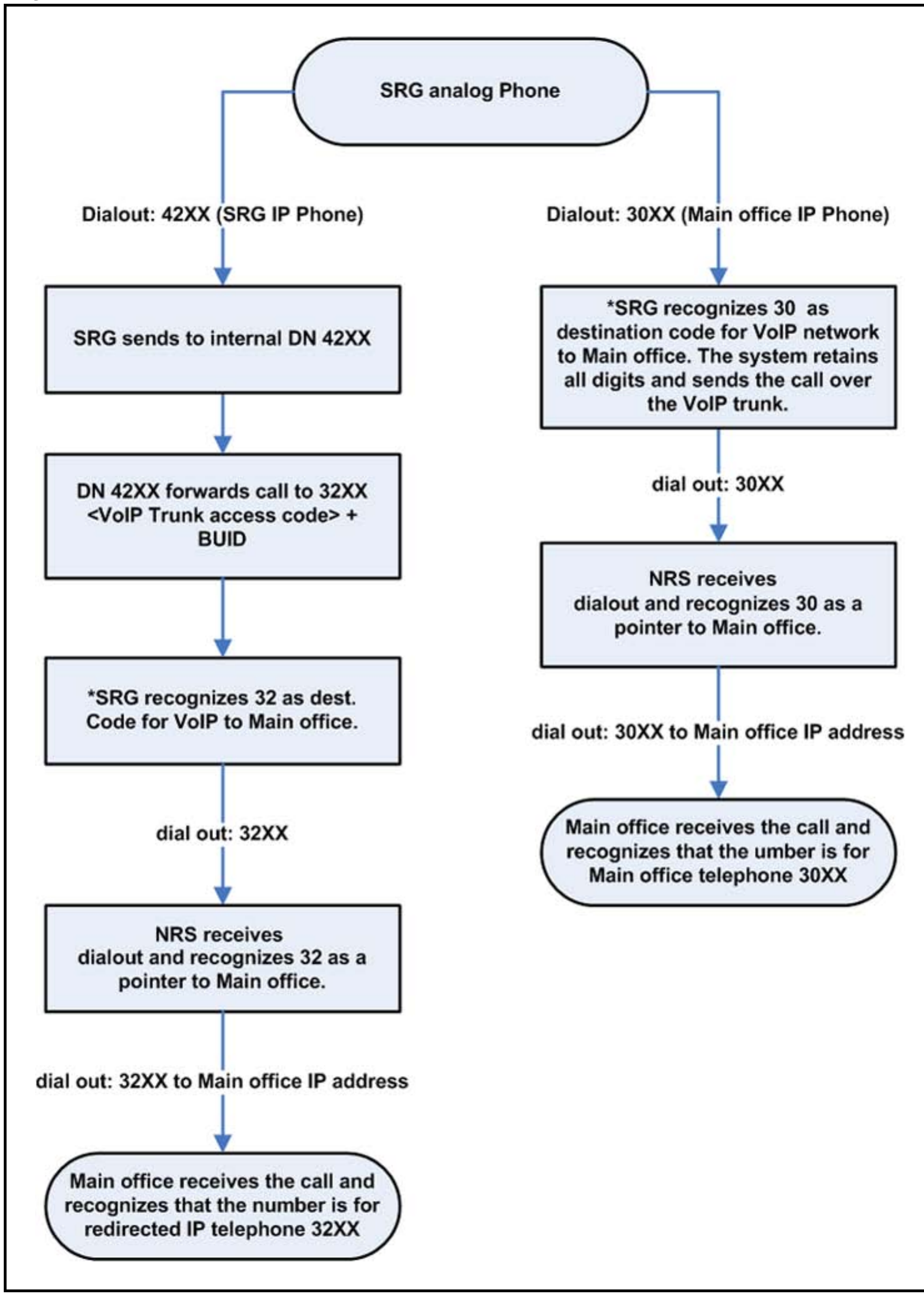
- Normal Mode: An SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office.

  In this scenario, the telephone registered to the SRG can either dial the SRG DN or the main office DN for the SRG IP Phone. In Local Mode, the SRG IP telephone is reached only with the SRG DN.

  In Normal Mode, the display on the IP Phone displays the main office DN (3xxx) for the IP Phone. In Local Mode, the SRG DN (4xxx) is displayed. The WAN is up: SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office (Normal Mode).

  Figure 21 "SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office" (page 62) shows this scenario.

Figure 21
SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office



- Local Mode: SRG IP Phones are registered at the SRG.

In this scenario, the WAN and the NCS are working. If the main office user dials the SRG DN (42xx) to call the IP Phone, the call goes through.

Configuration To configure the main office:

- Configure the ESN Control Block for CDP in LD 86.

```
> LD 86
REQ NEW
CUST 0
FEAT ESN
CDP YES
MXSC 50
NCDP 4
DLTN YES
```

- Configure the CDP Distant Steering Code (DSC) in LD 87.

```
> LD 87
REQ NEW
CUST 0
FEAT CDP
TYPE DSC
DSC 4
FLEN 4
RLI 12
```

To configure the NRS (H.323 Gatekeeper/SIP Redirect Server):

- Create CDP Domain: MO_BO_CDP.
- Create H.323/SIP endpoints: MO, BO.
- Create Numbering Plan entries in CDP Domain:
  - — Add 4 for endpoint BO.
  - — Add 30 for endpoint MO.

For information about configuring H.323 Gatekeeper/SIP Redirect Server, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

To configure the SRG:

- Set the BUID to the same number that was assigned for the TN by the main office.
- Set the main office VoIP Trunk Access code to 0.
- Do not assign a value to the main office trunk access code field.

For more information on configuring the main office and NRS, see *Basic Network Feature Fundamentals* (NN43001-579) and *IP Peer Networking Installation and Commissioning* (NN43001-313) . For more information on configuring the SRG, see*SRG 50 Configuration Guide* (NN40140-500) .

Uniform Dialing Plan

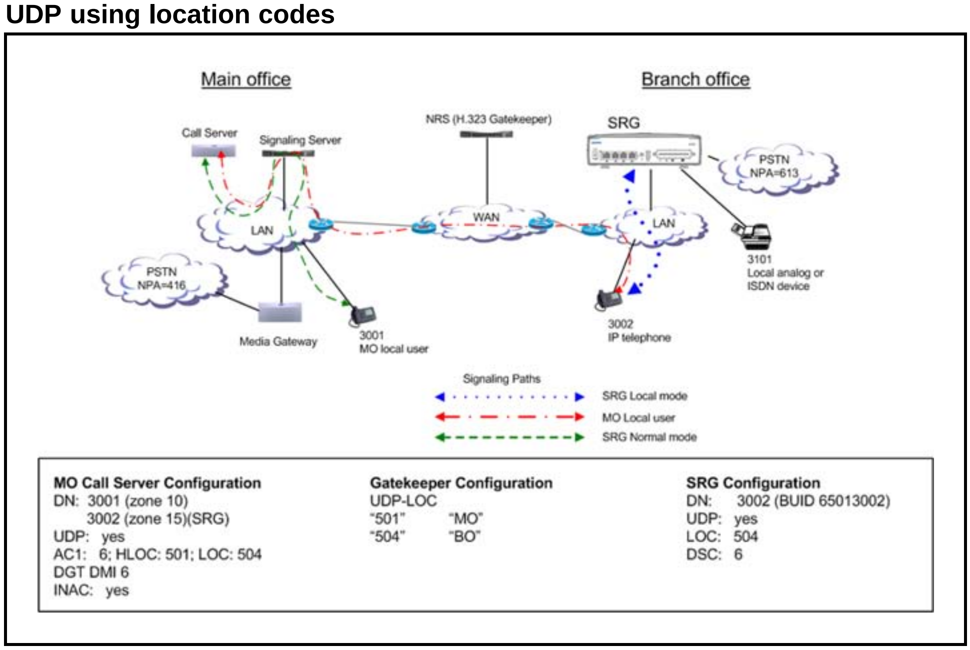
Overview

Figure 22 "UDP using location codes" (page 64) shows an example of a Uniform Dialing Plan (UDP) using location codes (Access Code + LOC + DN) configuration.

In this type of dialing plan, the DNs on the SRG do not need to be different from the BUID, since the location code (LOC) defines the unique node characteristic. Therefore, in this example:

- The SRG IP Phone has DN 3002 and BUID 3002. (The system adds the routing code and LOC code to the BUID).
- The local telephone has a DN of 3101.
- The main office has a telephone configured as TN 3001.
- On the main office, the AC1 steering code for the SRG is 6 and the LOC is 504.
- On the SRG, the destination code for the main is 6 and the LOC is 501.

Figure 22
UDP using location codes

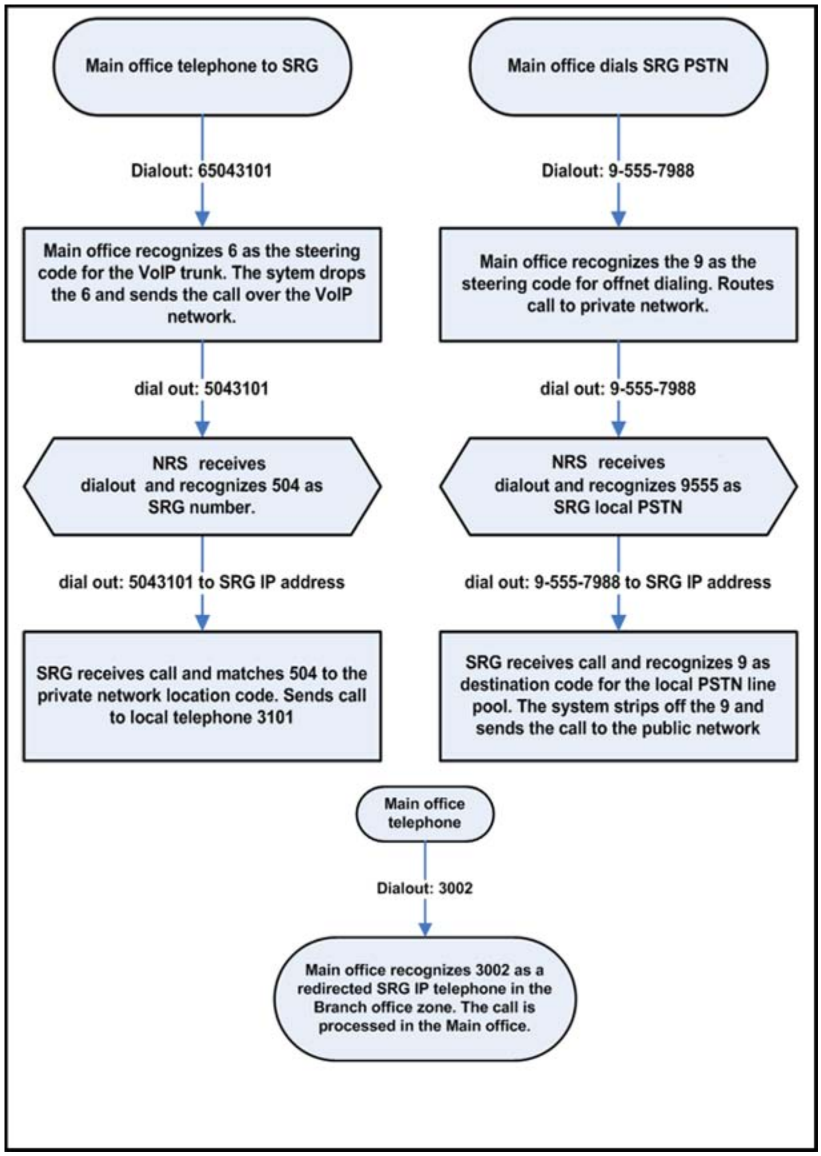


Call scenarios

This section describes how calls interact between the SRG and main office with UDP.

Calling from main office to the SRG and SRG PSTN, in Normal mode In this scenario, a telephone registered at the main office calls a telephone registered to the SRG, or makes a call over the PSTN through the SRG. Figure 23 "Calling from the main office to the SRG and SRG PSTN, in Normal Mode" (page 65) shows this scenario.

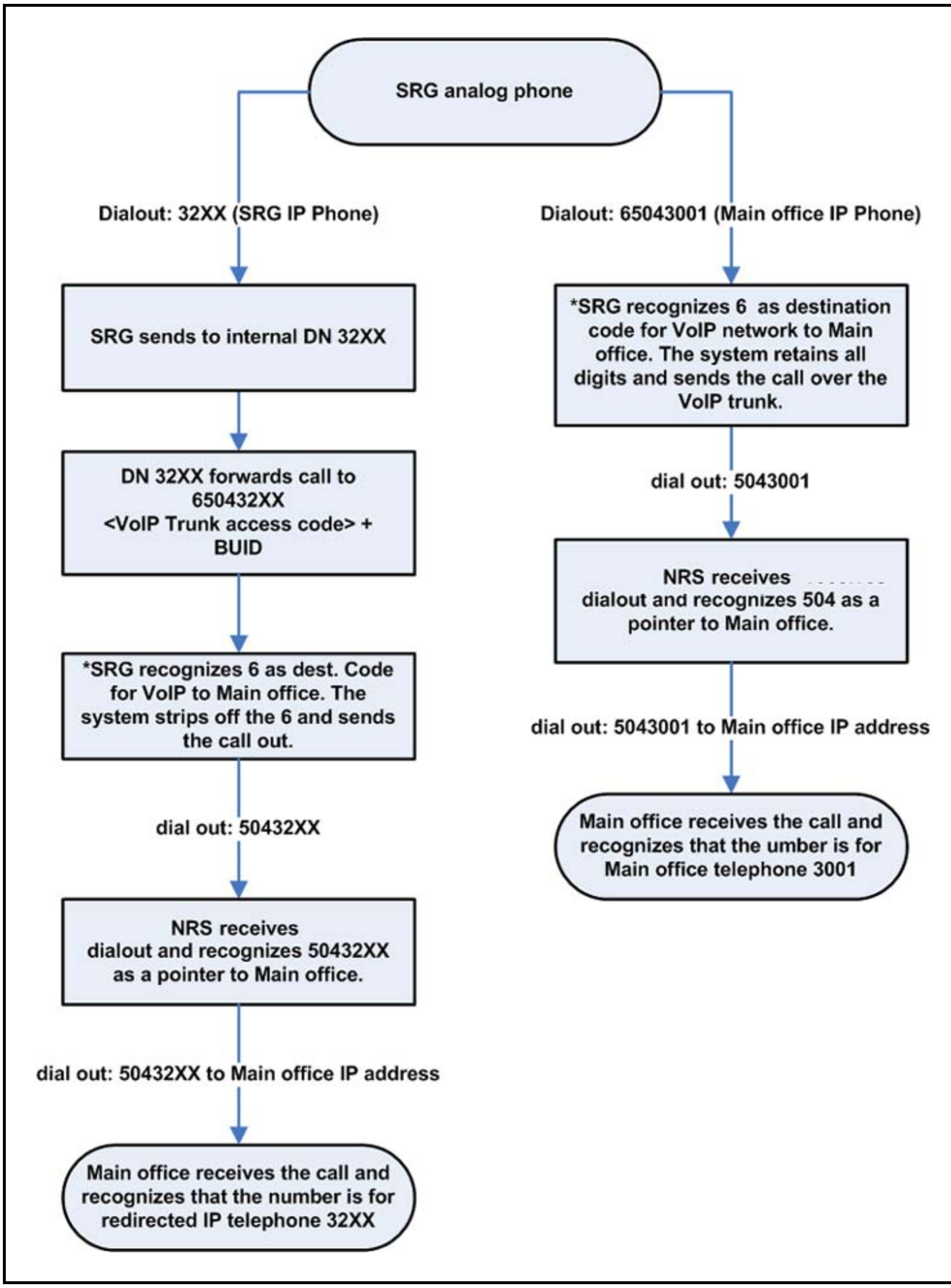
Figure 23
Calling from the main office to the SRG and SRG PSTN, in Normal Mode



Calling from the SRG to the main office, in Normal Mode In this scenario, a telephone registered at the SRG calls an SRG IP Phone and a main office IP Phone registered to the main office. The WAN is up. SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office (Normal Mode). Figure 24 "SRG analog

phone calls an SRG IP Phone and a main office IP Phone registered to the main office" (page 66)shows this scenario.

Figure 24
SRG analog phone calls an SRG IP Phone and a main office IP Phone registered to the main office



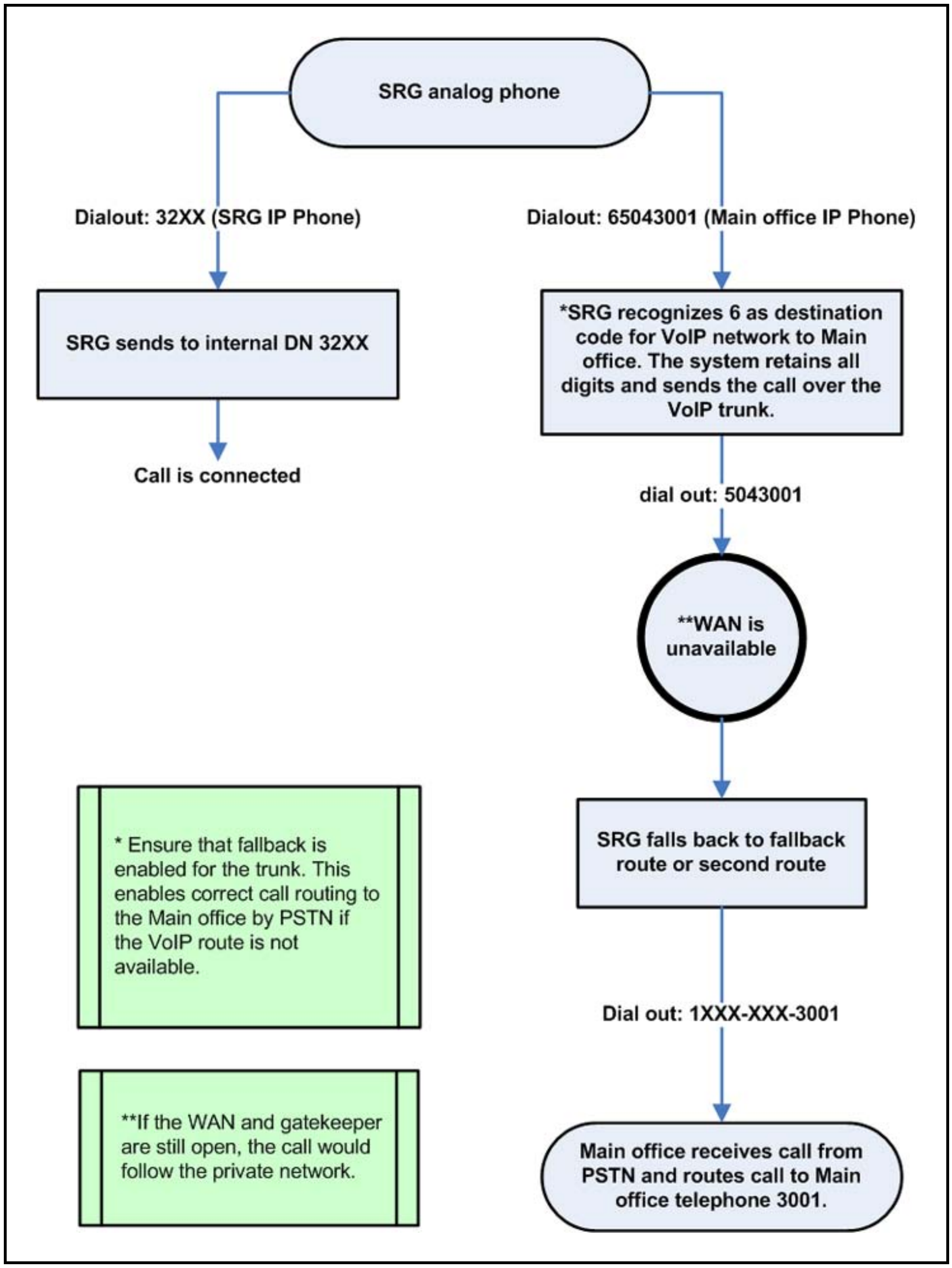
Calling in Local Mode In this scenario, the IP Phones at the SRG are in Local Mode because the WAN is down. The SRG IP telephones are reregistered to the SRG and call forward BUID is inactive on these telephones. These IP Phones are registered at the SRG, and call forward BUID is inactive on these telephones.

The inset shows a main office call to SRG telephones. The user must dial the SRG DN for the IP telephone (6002 instead of 3002). In this case, the user dialing is different in the following ways:

- DN 3001 can call DN 3002 by dialing 65043002, instead of 3002.
- DN 3101 can call DN 3002 by dialing 3002, instead of 65013002 dialed in Normal Mode.
- DN 3002 can call DN 3001 by dialing 65013001, instead of 3001 dialed in Normal Mode.
- DN 3002 can call DN 3101 by dialing 3101 instead of 65043101 dialed in Normal Mode.

The WAN is down. SRG analog phone calls an IP Phone and a main office IP Phone registered to the SRG (Local Mode). Figure 25 "SRG analog phone calls an IP Phone and a main office IP Phone registered to the SRG " (page 68) shows a call from the SRG to an SRG IP Phone and a main office IP Phone registered at the SRG.

Figure 25
SRG analog phone calls an IP Phone and a main office IP Phone registered to the SRG



Configuration examples

The following configurations are based on the examples provided in this section. For further information, see *Branch Office Installation and Commissioning* (NN43001-314) .

To configure the main office:

- Configure the ESN Control Block for UDP in LD 86.

```
> LD 86
REQ NEW
CUST 0
FEAT ESN
AC1 16
```

- Configure Digit Manipulation (DGT) in LD 86.

```
> LD 86
REQ NEW
FEAT DGT
DMI 6
DEL 3
```

- Configure the UDP Location Code (LOC) in LD 90.

```
> LD 90
REQ NEW
FEAT NET
TRAN AC1
TYPE LOC
LOC 504
FLEN 7
RLI 12
LDN 0
```

- Configure the UDP HLOC in LD 90.

```
FEAT NET
TRAN AC1
TYPE HLOC
HLOC 501
DMI 6
```

- Configure the HLOC in the Customer Data Block in LD 15.

```
> LD 15
REQ CHG
TYPE CDB
NET_DATA YES
ISDN YES
CLID YES
ENTRY <xx>
HLOC 501
```

- Configure the Virtual Trunk route in LD 16.

```
> LD 16
REQ NEW
TYPE RDB
CUST 00
ROUT 120
DES VTRKNODE51
TKTP TIE
VTRK YES
ZONE 101
NODE 51
PCID H323
ISDN YES
MODE ISLD
DCH 12
IFC SL1
INAC YES
```

To configure the NRS (H.323 Gatekeeper/SIP Redirect Server):

- Create H.323/SIP endpoints: MO, BO.
- Create Numbering Plan entries:
  - Choose type UDP-LOC.
  - Add 504 for endpoint BO.
  - Add 501 for endpoint MO.

For information about configuring H.323 Gatekeeper/SIP Redirect Server, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

To configure the SRG:

- Create route and destination code to main office.
- In the main office screen:
  - Set the type of number to ESN LOC.
  - The VoIP trunk access code field is empty.
  - Set the main office Access Code Length to 1.

  You can also include the LOC as the dial out when you configure the route for the VoIP line pool. This allows users to dial fewer numbers. For example, if 501 is configured as the dialout, and 6 is the destination code, the user could dial 6+<main office DN>. Once the system identifies the route (VoIP trunks) and drops the 6, it adds the LOC in front of the DN and dials <LOC>+<DN>. In the case of redirected IP Phones, the BUID is <destination code>+DN. The main office Access code length, in this circumstance, is set to 1.
- Dialing plan:

- — Set Type to UDP.
- — Set LOC to 504.

- Set the BUID on the IP Phones to <VoIP trunk destination code> + <LOC> + <DN>.

For more information on configuring the main office and NRS, see *Branch Office Installation and Commissioning* (NN43001-314) and *IP Peer Networking Installation and Commissioning* (NN43001-313) . For more information on configuring the SRG, see *SRG 50 Configuration Guide* (NN40140-500) .

Routing calls

SRG user call to an SRG PSTN

The SRG user telephone is registered at the main office. The SRG user telephones are physically located at the branch office, so routing of local PSTN calls back to the branch office is essential, even if they are registered with the main office.

Branch office behavior of the SRG user telephones at the main office is configured by setting branch office zone characteristics through LD 117 at the main office.

SRG PSTN to an SRG telephone (DID call)

If the DN is valid and can terminate, call termination at the branch office is treated differently for IP Phones and non-IP Phones, as follows:

- IP Phones—If the telephone is registered to the SRG (Local Mode), the call is terminated locally. If the telephone is not registered to the SRG (Normal Mode), the call is routed through a Virtual Trunk to the main office.
- Non-IP Phones—Calls are terminated locally (within the branch office).

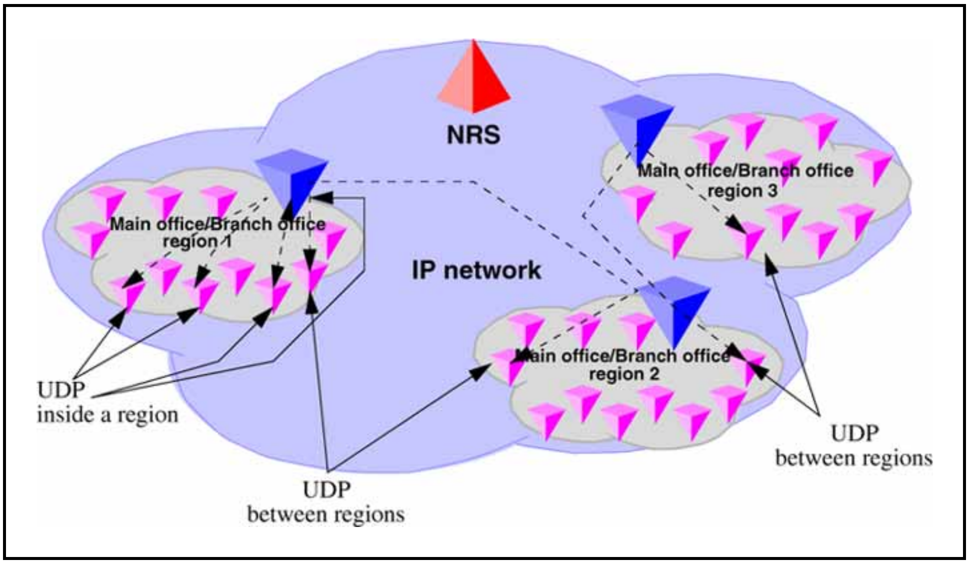
Network using Uniform Dialing Plan

The following section provides general network configuration for a network using UDP only.

Figure 26 "Scenario 1: UDP throughout the network" (page 72) shows two or more main offices with their branch offices, within a larger network. Callers within each main office/branch office region use UDP to place calls between systems. Callers also use UDP to place calls across the IP network to the other main office(s) and its (their) branch offices.

In a typical network, a full region uses a single Home Location Code (HLOC). However, it is also possible, where the number of users requires it, to have two or more codes, although using one for the main office and one for each branch office is unlikely at best.

Figure 26
Scenario 1: UDP throughout the network



Common details

In general, if an HLOC is shared between two or more systems, the provisioning at the main office gets more complex, unless all branch offices share HLOC with the main office. That is, if the main office has two or more HLOC, and one or more of these (but not necessarily the same one) is used by every branch office, then provisioning is relatively straight forward.

Table 10 "Configuration details for the general case" (page 72) describes the network configuration and the steps that a call takes during its setup.

Table 10
Configuration details for the general case

Region	Call progress steps	Configuration detail and call progress during call setup
1, 2, 3		UDP used for all calls within the region.
1, 2, 3		UDP used for region to region calls.
1, 2, 3		Prefixes for branch offices for regular calls are required for all branch offices. May have additional prefixes for E-911 calls, if required, or may share prefixes.

Table 10
Configuration details for the general case (cont'd.)

Region	Call progress steps	Configuration detail and call progress during call setup
1	1	All branch offices are provisioned at the NRS to route all outbound calls (from the branch office) through the main office. (NRS tandem configuration).
1	2	Main office sends all UDP calls to destinations that are not its own branch office to the NRS with unchanged dialled digits.
1	3	Main office sends all UDP calls to destinations that are its own branch office to the NRS with a specific gateway prefix in front of the dialled digits.
1	4	All branch offices delete the prefix and any LOC codes, and terminate the calls. May be to a local set or to a trunk.
2,3		Similar call setup steps take places for calls within region 2 and 3.

Differences when every branch office HLOC is shared with the main office

Table 11 "Provisioning details for this case" (page 73) shows the configuration when the branch office HLOC is shared with the main office.

Table 11
Provisioning details for this case

Region	Provisioning detail
1	Provisioning on the main office requires parsing to only normal LOC identification and HLOC deletion.
1	LOC values that are on branch offices may be provisioned as extended LOC (> 3 digit codes).
1	The DMI for the branch office LOC inserts a gateway routing prefix in front of the number.
2,3	Similar configuration, as above, applies to regions 2 and 3.

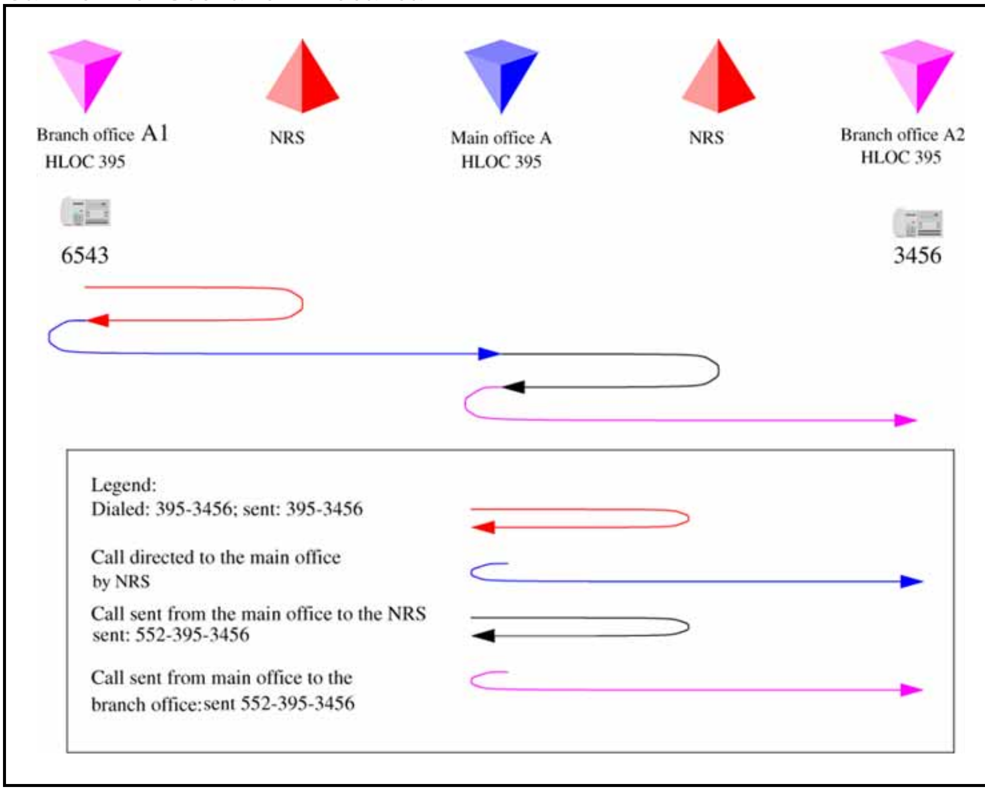
Call between two branch offices associated with the same main office

The following scenarios describe calls between two branch offices that belong to the same main office. The different scenarios described below vary in the manner in which the HLOC is architected; branch offices have same HLOC as the main office, branch offices have a different HLOC than the main office, and so on.

Every branch office HLOC is shared with the main office

In the following example, the HLOC of all the branch offices and the HLOC of the main office are all the same. See Figure 27 "Call flow for Scenario 1 - local call" (page 74).

Figure 27
Call flow for Scenario 1 - local call

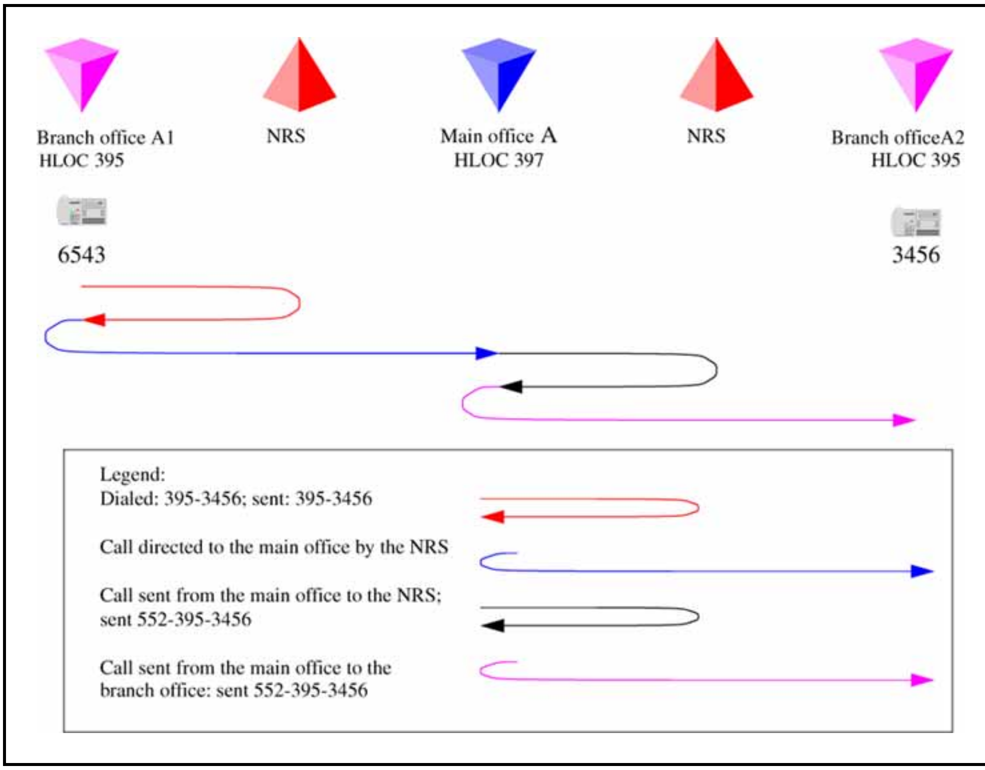


1. The branch office user dials 6-395-3456. The system transmits 395-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 395-3456 to the main office.
3. The main office determines that this is LOC 39534, to another branch office, with gateway routing prefix 552. The system inserts the prefix and transmits 552-395-3456 to the NRS. The NRS checks its provisioning, and determines that all calls to prefix 552 are to be sent to branch office A2; it directs the call to the branch office.
4. The main office sends the call to 552-395-3456 to the branch office. The branch office deletes the prefix and the HLOC, and rings set 3456.

No branch office HLOC is shared with the main office, but can be shared with another branch office

In this example, the HLOC of the branch offices are the same but the HLOC of the main office is different. See Figure 28 "Call flow for Scenario 1 - local call" (page 75).

Figure 28
Call flow for Scenario 1 - local call

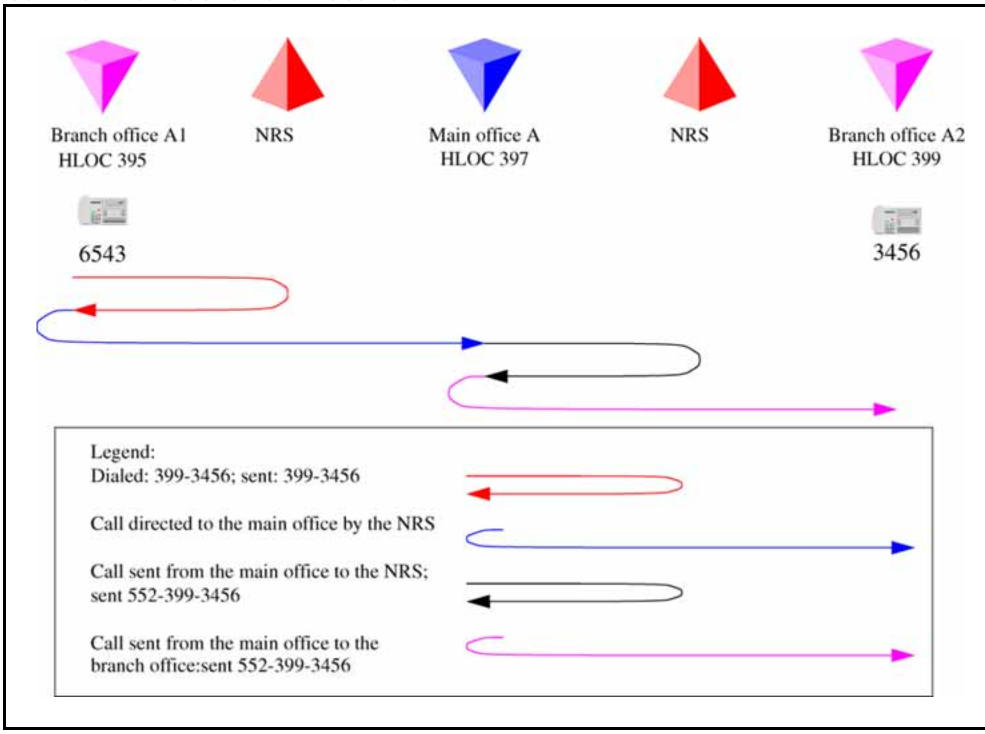


1. The branch office user dials 6-395-3456. The system transmits 395-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 395-3456 to the main office.
3. The main office determines that this is LOC 39534 to another branch office, with gateway routing prefix 552. The system inserts the prefix and transmits 552-395-3456 to the NRS. The NRS checks its provisioning, and determines that all calls to prefix 552 are to be sent to branch office A2; it directs the call to the branch office.
4. The main office sends the call to 552-395-3456 to the branch office. The branch office deletes the prefix and the HLOC and rings set 3456.

No branch office HLOC is shared with the main office or another branch office

In this example, the HLOC is unique between all the branch offices and the main office. See Figure 29 "Call flow for Scenario 1- local call" (page 76).

Figure 29
Call flow for Scenario 1- local call



1. The branch office user dials 6-395-3456. The system transmits 399-3456 to the branch office user dials 6-399-3456. NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 399-3456 to the main office.
3. The main office determines that this is to another branch office, with office prefix 552. The system inserts the prefix and transmits 552-399-3456 to the NRS. The NRS checks its provisioning, and determines that all calls to prefix 552 are to be sent to branch office A2; it directs the call to the branch office.
4. The main office sends the call to 552-399-3456 to the branch office. The branch office deletes the prefix and the HLOC, and rings set 3456.

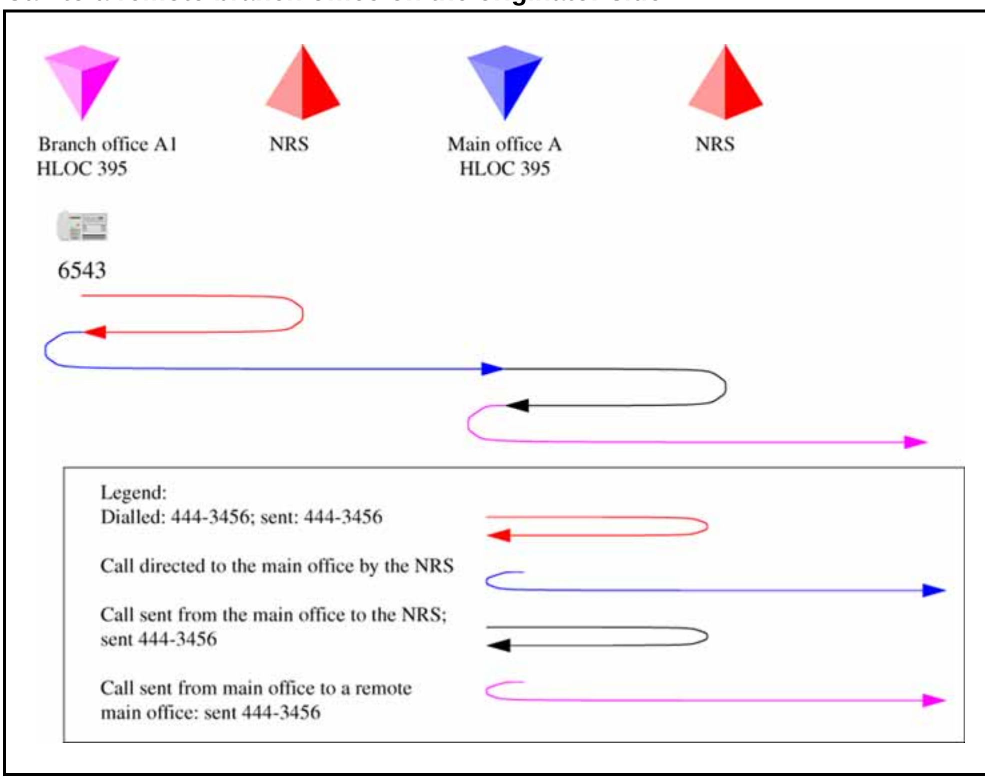
Call between branch offices associated with different main office

The following scenarios describe calls between two branch offices that belong to different main offices. Note that the different scenarios described below vary in the manner in which the HLOC is architected; branch offices have same HLOC as the main office, branch offices have a different HLOC than the main office, and so on.

Every branch office HLOC is shared with the main office

Figure 30 "Call to a remote branch office on the originator side " (page 77) shows the first half of the call setup (the originator side is side A). In this example, the branch office and the main office share the same HLOC. Figure 31 "Call to remote branch office on the destination side" (page 78) shows the second half of the call (the terminating side is side B).

Figure 30
Call to a remote branch office on the originator side

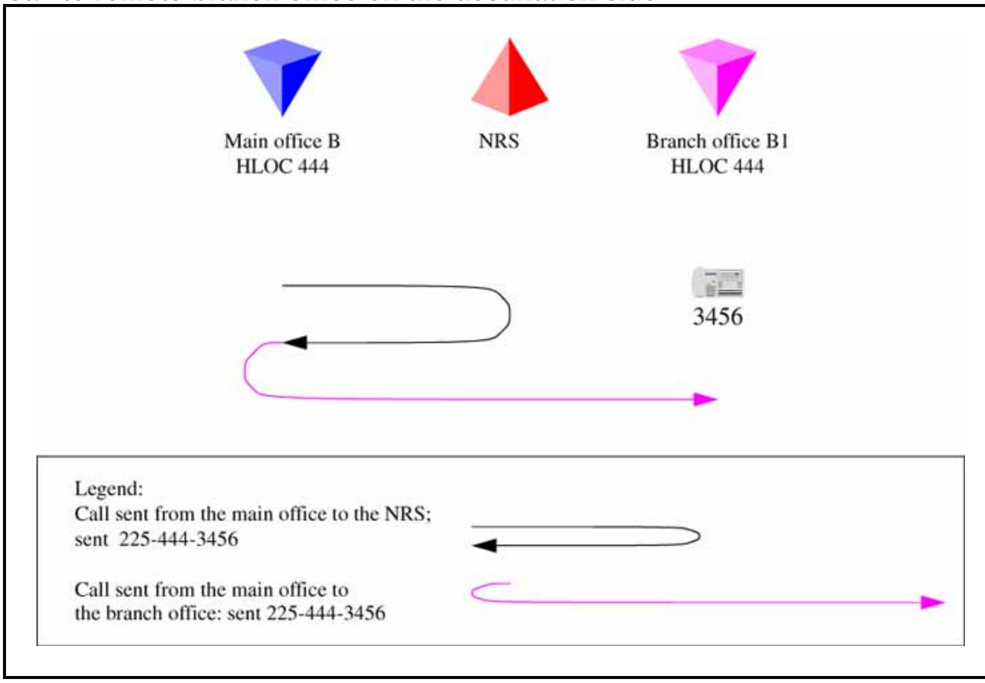


1. The branch office user dials 6-444-3456. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and

determines that all calls are to be sent to the main office; it directs the call to the main office.

2. The branch office sends the call to 444-3456 to the main office.
3. The main office determines that this is to another main office. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to main office B.

Figure 31
Call to remote branch office on the destination side

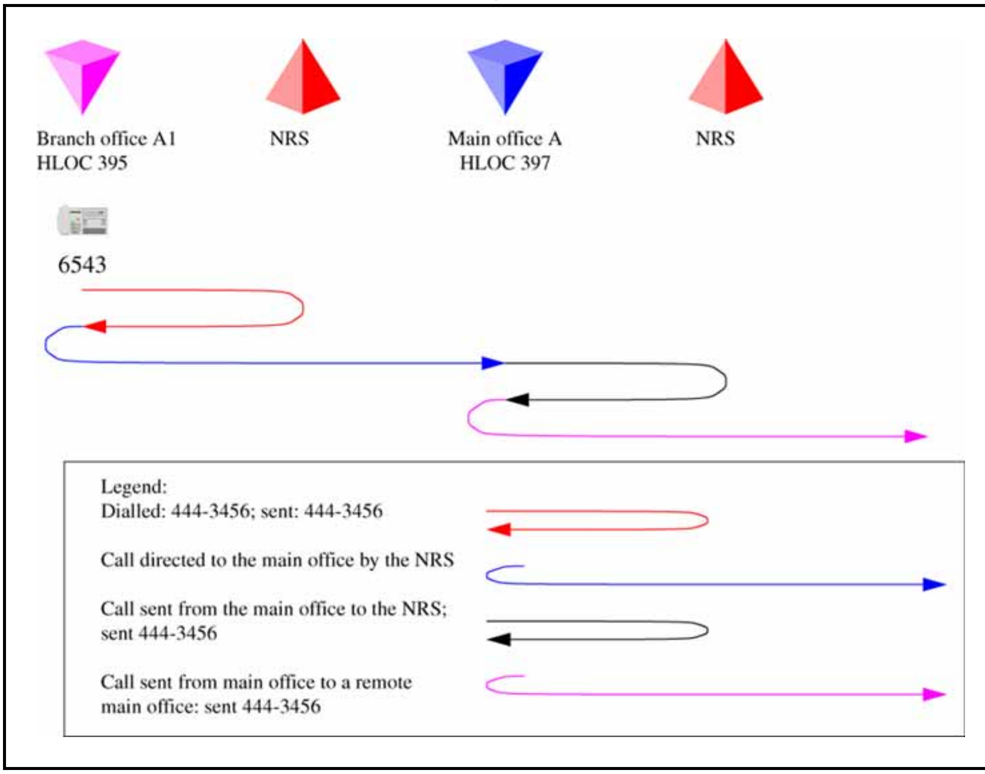


1. Main office B determines that this is to LOC 44434, which is a local branch office with prefix 225. The system transmits 225-444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to branch office B1.
2. The main office sends the call to 225-444-3456 to the branch office. The branch office deletes the prefix, discovers the call is to its HLOC 444, deletes the HLOC, and rings set 3456.

No branch office HLOC is shared with the main office, but can be shared with another branch office

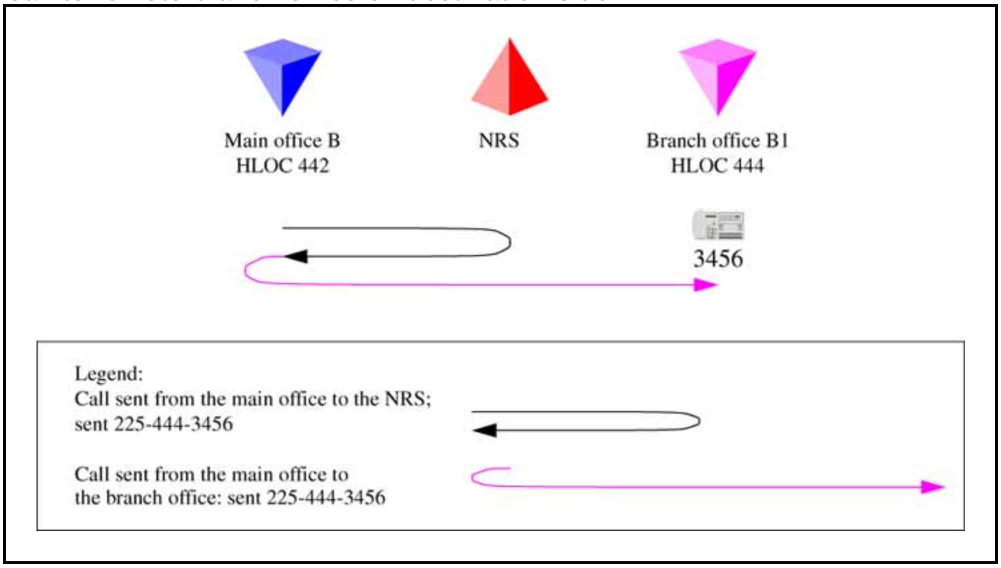
Figure 32 "Call to remote branch office on the originator side" (page 79)shows the first half of the call (originator side of the call). Figure 33 "Call to remote branch office on destination side" (page 80)shows the second half of the call (destination side of the call).

Figure 32
Call to remote branch office on the originator side



1. The branch office user dials 6-444-3456. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 444-3456 to the main office.
3. The main office determines that this is to another main office. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to main office B.

Figure 33
Call to remote branch office on destination side

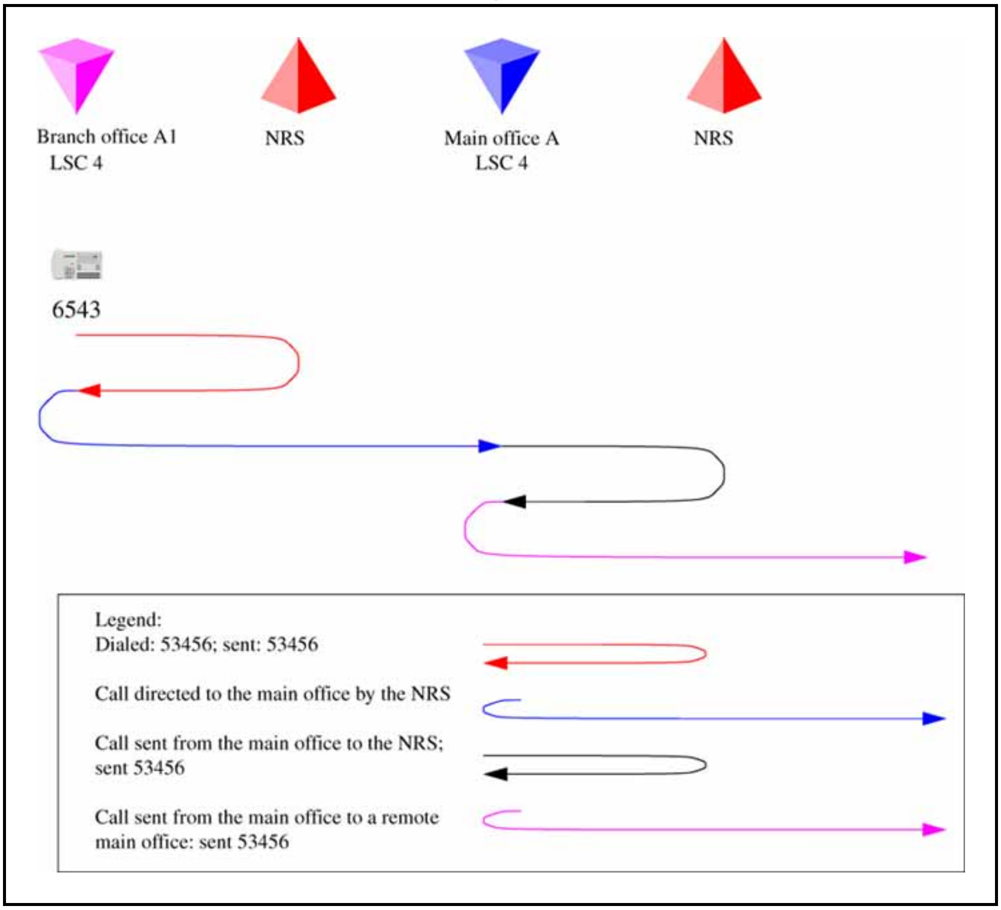


1. Main office B determines that this LOC plus digits is to a local branch office with prefix 225. (If sharing this LOC with another branch office, the extended LOC is 44434.) The system transmits 225-444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to branch office B1.
2. The main office sends the call to 225-444-3456 to the branch office. The branch office deletes the prefix, and the HLOC, and rings set 3456.

No branch office HLOC is shared with the main office or another branch office

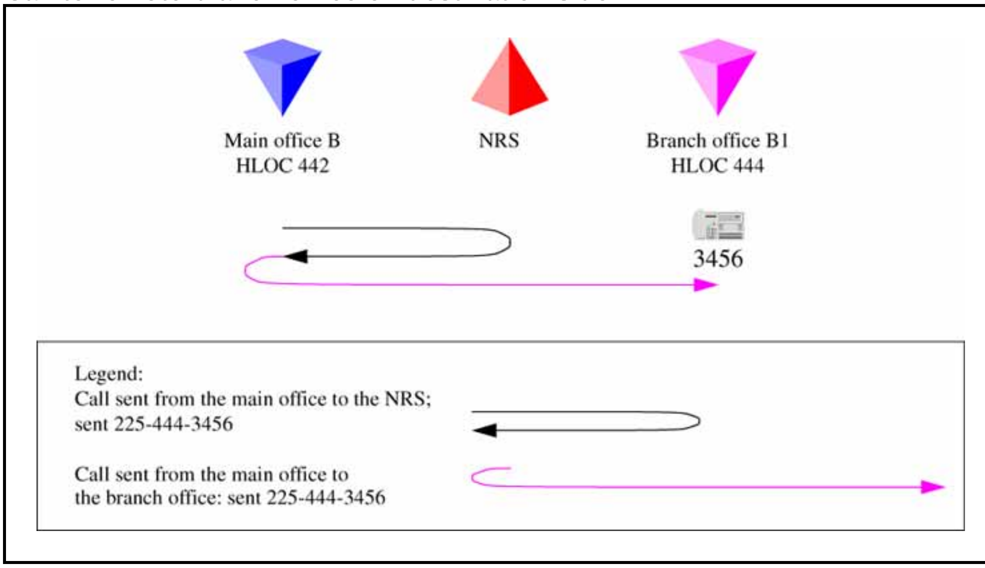
The following example shows a call between two branch offices. In this example, the HLOC is unique between the main office and branch office. Figure 34 "Call to remote branch office on the originator side" (page 81)shows the first half of the call (originator side of the call). In Figure 35 "Call to remote branch office on destination side" (page 82) shows the second half of the call (destination side of the call).

Figure 34
Call to remote branch office on the originator side



1. The branch office user dials 6-444-3456. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 444-3456 to the main office.
3. The main office determines that this is to another main office. The system transmits 444-3456 to the NRS. NRS checks its provisioning, and determines that this call goes to main office B.

Figure 35
Call to remote branch office on destination side



1. Main office B determines that LOC 444 is to a local branch office with prefix 225. The system transmits 225-444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to branch office B1.
2. The main office sends the call to 225-444-3456 to the branch office. The branch office deletes the prefix, discovers the call is to its HLOC, deletes the HLOC, and rings set 3456.

Summary of provisioning procedures for Tandem Bandwidth Management

Use Procedure 3 "Provisioning Tandem Bandwidth Management" (page 84) to provision the network.

Step	Action
1	Enter the main office Gateway endpoint identifier in the Tandem Endpoint field for each branch office gateway configured on the NRS. This provides tandeming for outbound calls from a branch office through its main office. See Step 1.
2	Plan the gateway routing prefixes, if not already done. At least one prefix is needed for each branch office, although any branch offices that have a prefix for ESA 911 calls does not necessarily require another. (These prefixes are Special Number [SPN] entries.) See Step 2.
3	Provision the NRS to send all calls to a LOC without a gateway routing prefix to the main office of that LOC, or to the main office

which provides service for the branch office using the LOC. See Step 3.

4 Provision the NRS to send all calls to a LOC with a gateway routing prefix to the branch office directly. Using the gateway routing prefix and the Type of Number of SPN, the entries can be differentiated from the normal LOC easily. See Step 4.

5 Provision the main office with the DGT table DMI to insert the prefixes and set the Type of Number correctly. Create RLB RLI entries to use these DMI for the VTRK route(s). One RLI per branch office will be the minimum requirement. Note that calls from remote systems will typically have the HLOC prefix, so this is defined here. See Step 5.

6 Provision the main office with CDP DSC (mapped by the RLI into Location Codes) sufficient to uniquely identify all of its branch offices (using extended location codes, if required); use the RLI index defined for each branch office as the RLI value of the LOC definition. This is the route to the branch office. See Step 6.

7 Provision the main office and branch office with a home location code (HLOC) or multiple codes to terminate all calls that should terminate on this system. See Step 7.

8 Provision the main office to send all other LOC to the IP network without prefixes. These are going to a remote main office. See Step 8.

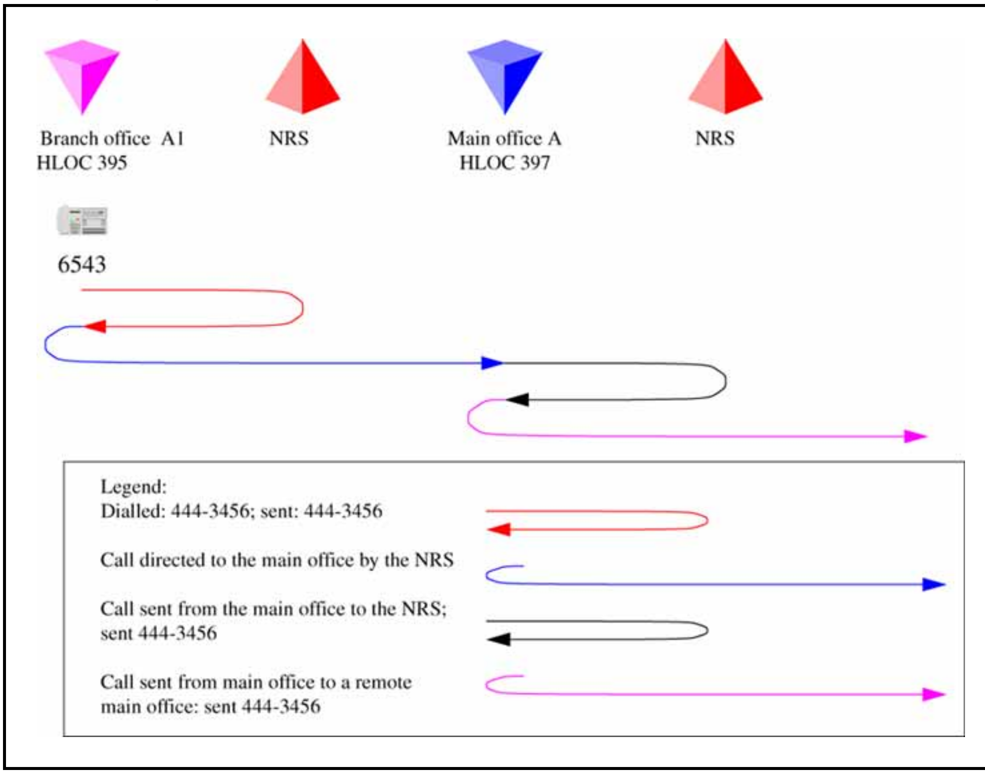
9 Provision the branch office with a terminating RLI with a DMI to delete the LOC prefixes. See Step 9.

--End--

Provisioning Example of Tandem Bandwidth Management

Figure 36 "Provisioning example" (page 84) shows an example of the network configuration.

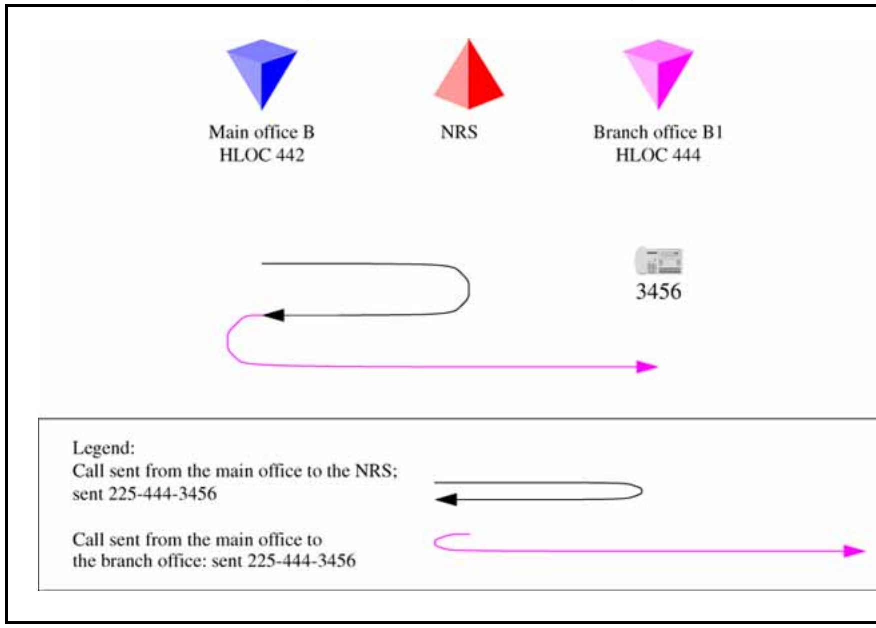
Figure 36
Provisioning example



Procedure 3
Provisioning Tandem Bandwidth Management

Step	Action
1	Enter the main office Gateway endpoint identifier in the Tandem Endpoint field for each branch office GW configured on the NRS. This provides tandeming for outbound calls from a branch office through its main office. Figure 37 "Tandem endpoint configuration in Element Manager" (page 85) shows the tandem endpoint configuration in Element Manager.

Figure 37
Tandem endpoint configuration in Element Manager



2 Plan the gateway routing prefixes. At least one prefix is needed per branch office, although any branch offices that have a prefix for ESA 911 calls does not necessarily require another. (These prefixes will be SPN - Special Number - entries if you are using ESA 911. In the example these are LOC codes because *network 911* is not being used.) In our example the Branch office prefixes are 741 (branch office B) and 742 (branch office A).

3 Provision the NRS to send all calls to a LOC without a gateway routing prefix to the main office of that LOC, or to the main office which provides service for the branch office using the LOC. In our example the NRS is provisioned with 841 (for main office B) and 842 (for main office A).

4 Provision the NRS to send all calls to a LOC with a gateway routing prefix to the branch office directly. Using the gateway routing prefix and the Type of Number as used (LOC or SPN), the entries can be differentiated from the normal LOC easily. In our example the NRS is provisioned with 741-841 at branch office B and 742-842 for branch office A.

5 Provision the main office with the DGT table DMI to insert the prefixes and set the Type of Number correctly. Create RLB RLI entries to use these DMI for the VTRK route(s). One RLI per branch office will be the minimum requirement. Note that calls from remote systems will typically have the HLOC prefix, so this is defined here

Table 12 "Main office B DMI and RLI provisioning for calls in branch office B" (page 86) lists main office B DMI and RLI provisioning.

Table 12
Main office B DMI and RLI provisioning for calls in branch office B

Create a DMI	Create an RLI
LD 86	LD 86
REQ new	REQ new
CUST 0	CUST 0
FEAT dgt	FEAT rlb
DMI 50	RLI 50
DEL 0	ENTR 0
ISPN no	LTER no
INST 741841	ROUT 71
CTYP loc	DMI 50

6 Provision the main office with CDP DSC (mapped by the RLI into Location Codes) sufficient to uniquely identify all of its branch offices (using extended location codes, if required); use the RLI index defined for each branch office as the RLI value of the LOC definition. This is the route to the branch office.

Table 13 "Main office B LOC provisioning for LOC 741 841" (page 86) lists main office B LOC provisioning.

Table 13
Main office B LOC provisioning for LOC 741 841

Create a CDP mapped to the LOC:
LD 87
REQ NEW
CUST 0
FEAT CDP
TYPE DSC
DSC 4030
FLEN 4
RLI 50
Create a CDP mapped to the LOC:
LD 87

7 Provision the main office and branch office with a home location code (HLOC) or multiple codes to terminate all calls that should terminate on this system.

Table 14 " Main office B and branch office B" (page 87) lists main office and branch office HLOC provisioning.

Table 14
Main office B and branch office B

Create a DMI	Create an HLOC
LD 86	LD 90
REQ new	REQ new
CUST 0	CUST 0
FEAT dgt	FEAT net
DMI 61	TRAN ac1
DEL 3	TYPE hloc
ISPN no	HLOC 841
	DMI 61

Repeat the above for all the main offices and branch offices.

8 Provision the main office to send all other LOC to the IP network without prefixes. These are going to a remote main office.

Table 15 "Main office B LOC provisioning for LOC to remote main office system " (page 87) lists main office B LOC provisioning for LOC to remote main office. The Main Office A is LOC 842.

Table 15
Main office B LOC provisioning for LOC to remote main office system

Create an RLI	Create a LOC
LD 86	LD 90
REQ new	REQ NEW
CUST 0	CUST 0
FEAT rlb	FEAT NET
RLI 51	TRAN AC1
ENTR 0	TYPE LOC
LTER no	LOC 842
ROUT 71	FLEN 7
	RLI 51

Table 16 "Main office A LOC provisioning for LOC to remote main office systems " (page 88) lists main office A LOC provisioning for LOC to the remote office. The Main office B is LOC 841

Table 16
Main office A LOC provisioning for LOC to remote main office systems

Create an RLI	Create a LOC
LD 86	LD 90
REQ new	REQ NEW
CUST 0	CUST 0
FEAT rlb	FEAT NET
RLI 71	TRAN AC1
ENTR 0	TYPE LOC
LTER no	LOC 841
ROUT 75	FLEN 7
	RLI 71

9 Provision the branch office with a terminating RLI with a DMI to delete the LOC prefixes.

Table 17
Branch office terminating RLI provisioning

Create a DMI	Create an HLOC
LD 86	LD 90
REQ new	REQ NEW
CUST 0	CUST 0
FEAT dgt	FEAT net
DMI 61	TRAN ac1
DEL 6	TYPE hloc
ISPN no	HLOC 741
	DMI 61

--End--

Network using mixed Coordinated Dialing Plan and Uniform Dialing Plan

The following section provides general details of the network setup. shows an example of a mixed network configuration.

Figure 38
UDP between main offices and CDP inside the main office region

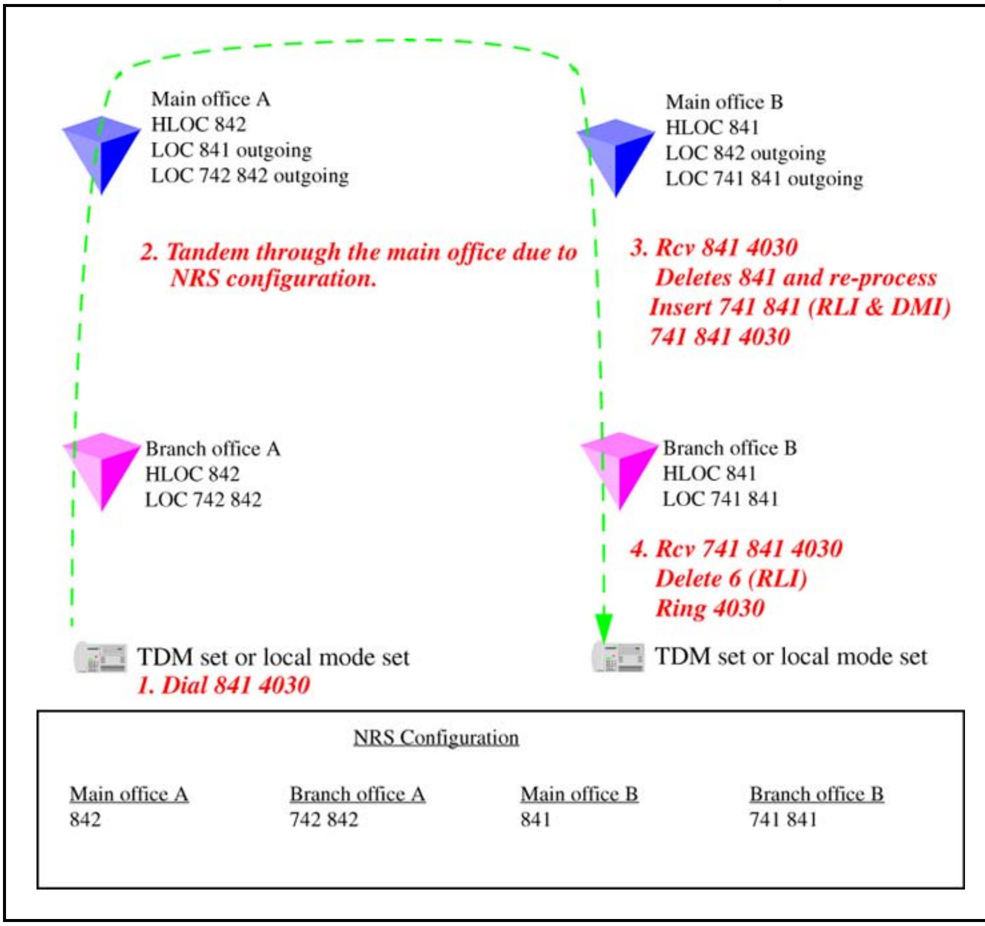


Table 18 "Provisioning details for this case" (page 89) lists provisioning details for a mixed network.

Table 18
Provisioning details for this case

Region	Provisioning detail
1, 2, 3	CDP used for all calls within the region.
1, 2, 3	UDP used for region to region calls.
1, 2, 3	Prefixes for branch offices for regular calls not required. May still have prefixes for E-911 calls, if required.
1	All branch offices are provisioned at the NRS to route all calls through the main office.
1	Main office sends all UDP calls to destinations that are not its own branch office to the NRS with unchanged dialled digits.

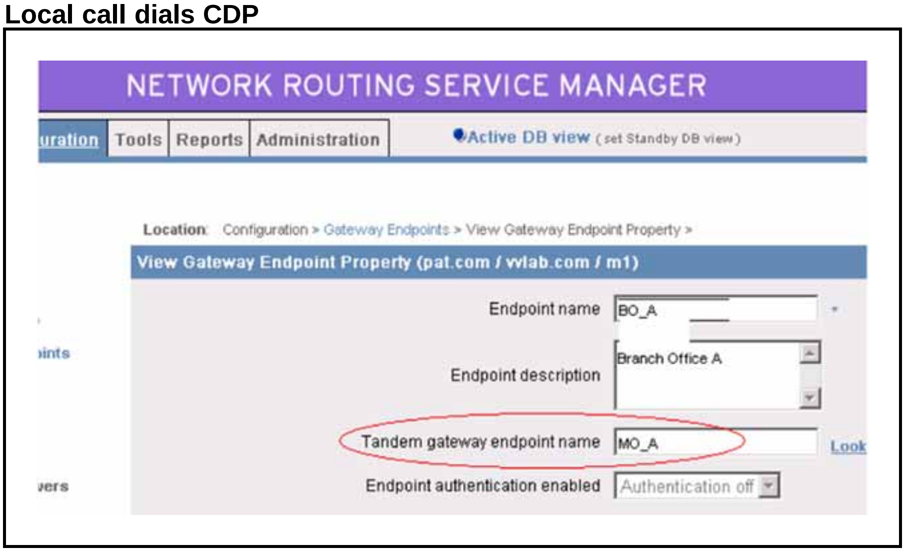
Table 18
Provisioning details for this case (cont'd.)

Region	Provisioning detail
1	Main office sends all UDP calls to destinations that are its own branch office to the NRS after deleting the HLOC and converting to CDP.
2,3	Similar configuration, as above, applies to regions 2 and 3.

Call between two local branch offices

Figure 39 "Local call dials CDP" (page 90) shows the NRS Configuration web page in Element Manager.

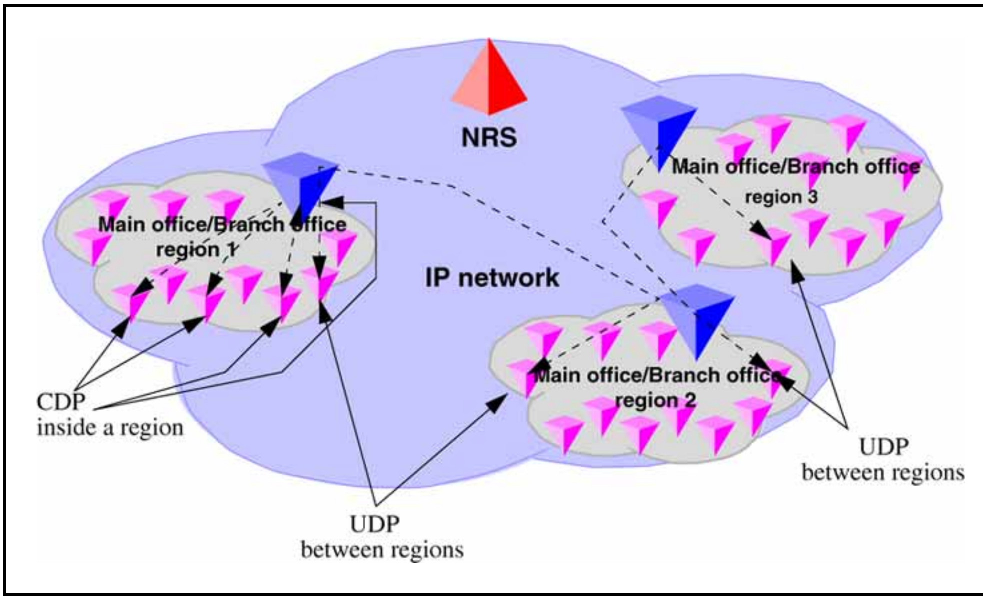
Figure 39
Local call dials CDP



1. The branch office user dials 3456 (CDP). The system transmits 3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 3456 to the main office.
3. The main office determines that this is to another branch office. The system transmits 3456 to the NRS. The NRS checks its provisioning, and determines that all calls to 3456 in this CDP domain are to be sent to branch office A2; it directs the call to the branch office.
4. The main office sends the call to 3456 to the branch office. The branch office rings set 3456.

Abnormal case - calls originating using UDP, but terminating using CDP

Figure 40
Call flow for Scenario 2 - local call dial UDP

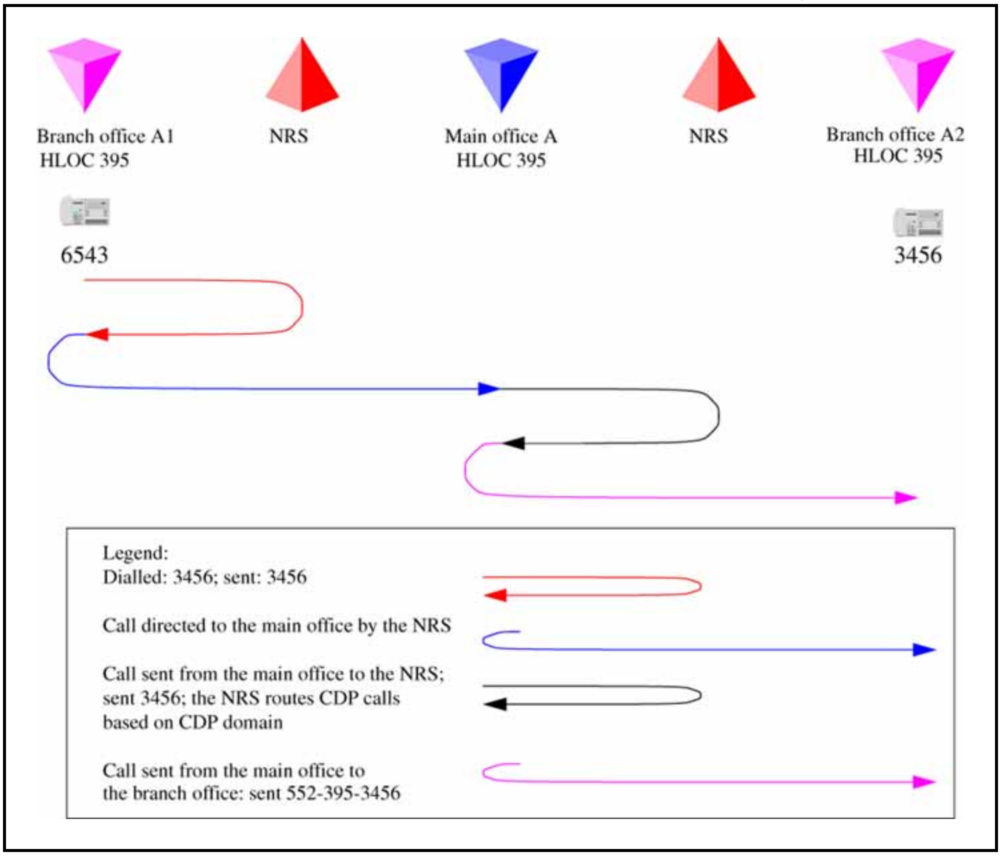


1. The branch office user dials 6-395-3456. The system transmits 395-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 395-3456 to the main office.
3. The main office determines that this is to another branch office, using CDP. The system deletes the HLOC and transmits 3456 to the NRS. The NRS checks its provisioning, and determines that all calls to 3456 from this CDP region are to be sent to branch office A2; it directs the call to the branch office.
4. The main office sends the call to 3456 to the branch office. The branch office rings set 3456.

Call between branch offices associated with different main offices

In the following diagram, the first half of the call is shown (the originator side of the call).

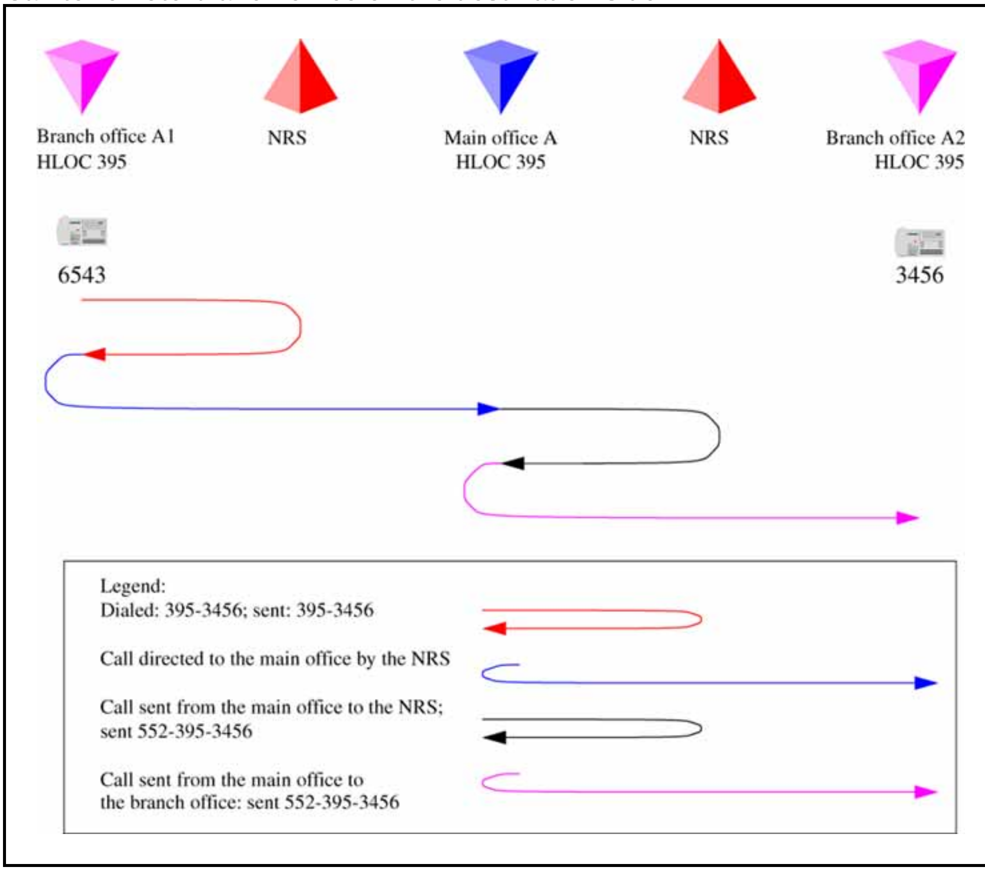
Figure 41
Call flow for Scenario 2 - local call to remote branch office (originator side)



1. The branch office user dials 6-444-3456. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 444-3456 to the main office. The main office determines that this is to another main office. The system transmits 444-3456 to the NRS. The NRS checks its provisioning, and determines that this call goes to main office B.

Figure 42 "Call to remote branch office on the destination side" (page 93) shows the second half of the call (destination side of the call).

Figure 42
Call to remote branch office on the destination side



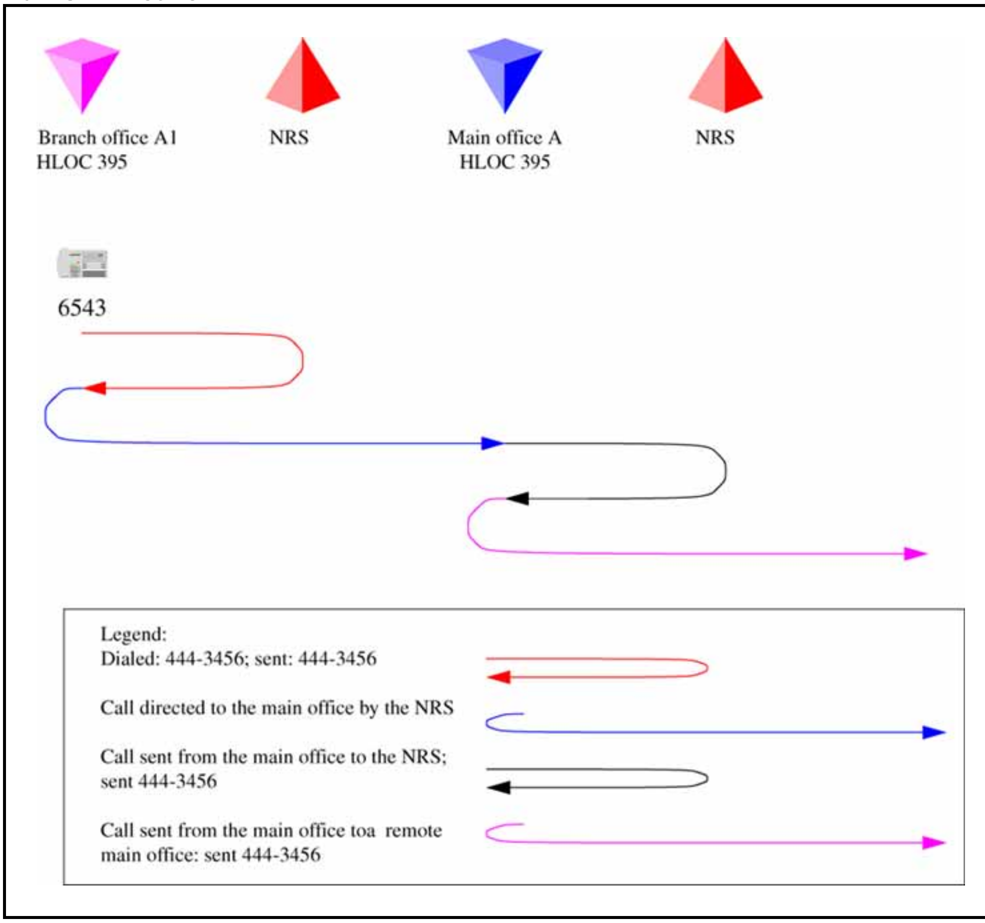
1. The main office B deletes the HLOC, and determines that this is to a local branch office. The system transmits 3456 to the NRS. The NRS checks its provisioning, and determines that for this CDP region this call goes to branch office B1.
2. The main office sends the call to 3456 to the branch office. The branch office rings set 3456.

Network using Coordinated Dialing Plan

The following section provides general details of network setup.

The following diagram shows a full CDP network configuration.

Figure 43
Full CDP network



The following table lists the provisioning details for a full CDP network.

Table 19
Provisioning details for this case

Region	Provisioning detail
1, 2, 3	CDP used for all calls within the region.
1, 2, 3	CDP used for region to region calls.
1, 2, 3	All CDP numbers must be sufficiently long to allow unique termination of the calls. That is, every main office/branch office region requires its own LSC to ensure that all numbers are unique.
1, 2, 3	Prefixes for branch offices for regular calls are required. May have additional prefixes for E-911 calls, if required, or may share prefixes.
1	All branch offices are provisioned at the NRS to route all calls through the main office.

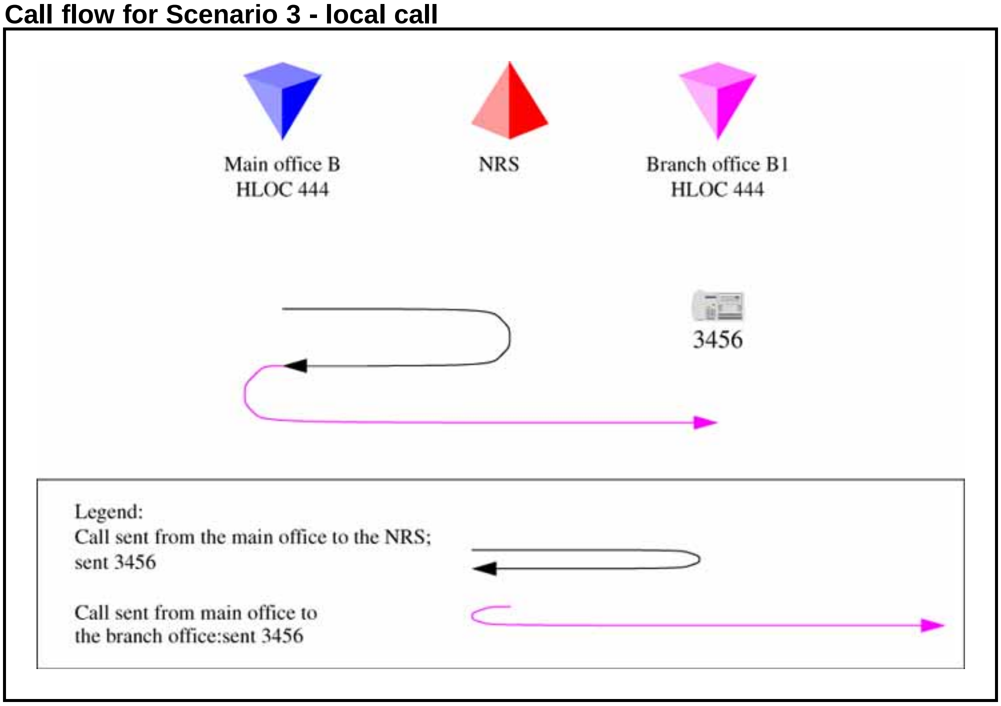
Table 19
Provisioning details for this case (cont'd.)

Region	Provisioning detail
1	Main office sends all CDP calls to destinations that are not its own branch office to the NRS with unchanged dialled digits.
1	Main office sends all CDP calls to destinations that are its own branch office to the NRS with a specific gateway prefix in front of the dialled digits.
1	All branch offices delete the prefix and terminate the calls. May be to a local set or to a trunk.
2,3	Similar configuration, as above, applies to regions 2 and 3.

Call between two local branch offices

The following diagram shows the call flow of a call between two local branch offices.

Figure 44
Call flow for Scenario 3 - local call



1. The branch office user dials 43456. The system transmits 43456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 43456 to the main office.
3. The main office determines that this is to another branch office, with office prefix 552. The system inserts the prefix and transmits

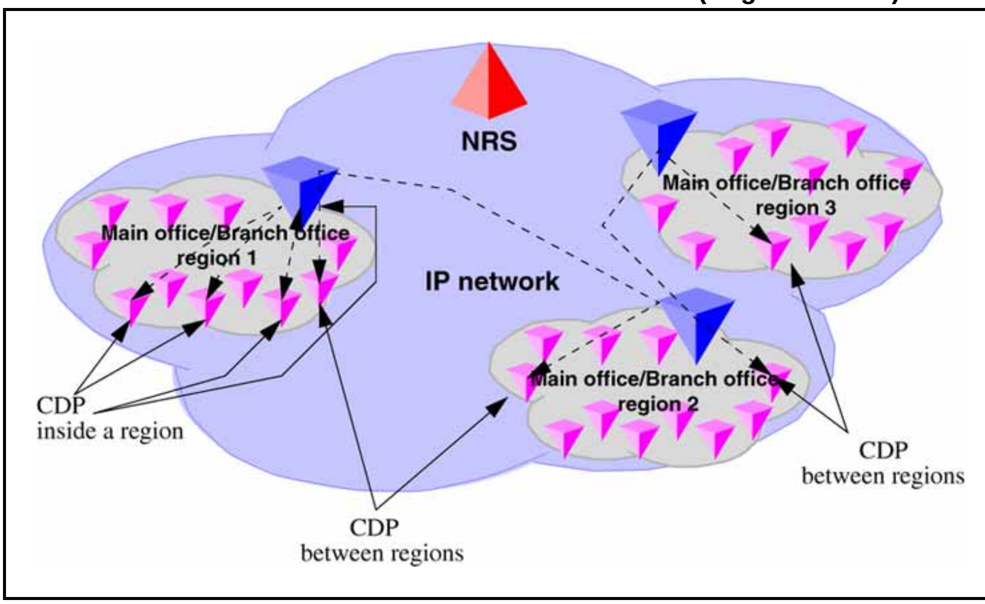
552-43456 to the NRS. The NRS checks its provisioning, and determines that all calls to prefix 552 are to be sent to branch office A2; it directs the call to the branch office.

4. The main office sends the call to 552-43456 to the branch office. The branch office deletes the prefix and LSC 4, and rings set 3456.

Call between branch offices associated with different main offices

In the following diagram, the first half of the call is shown (originator side of the call).

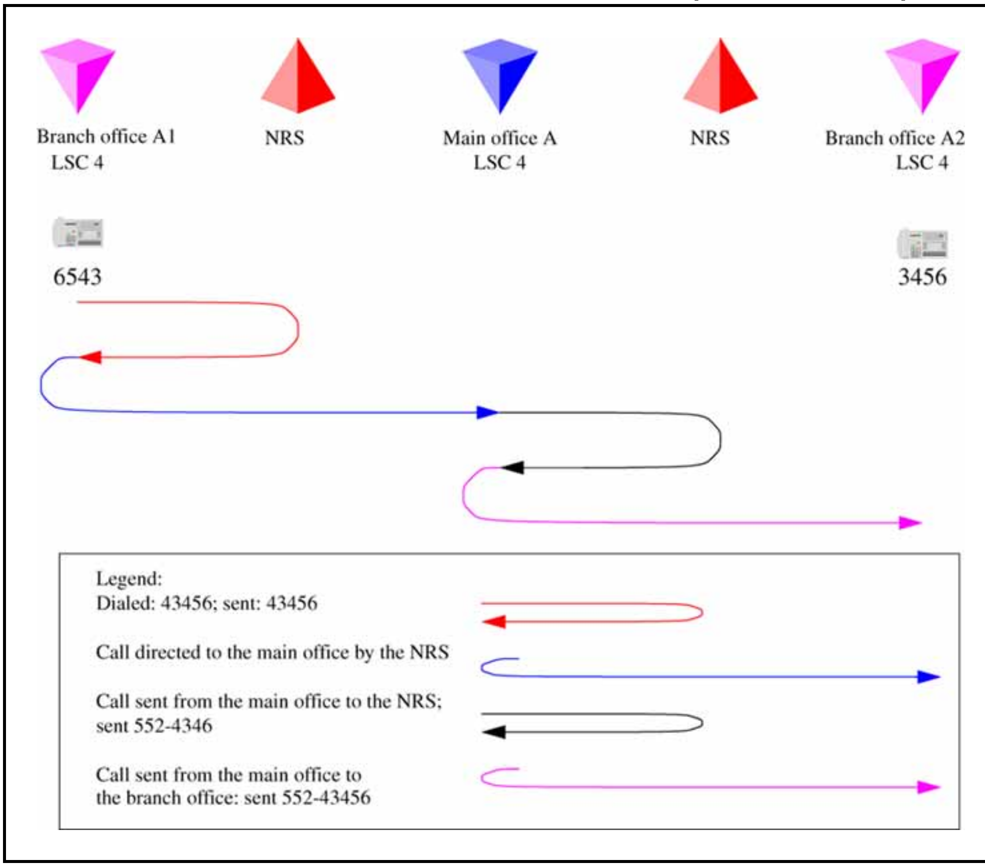
Figure 45
Call flow for Scenario 3 - calls to remote branch office (originator side)



1. The branch office user dials 53456. The system transmits 53456 to the NRS. The NRS checks its provisioning, and determines that all calls are to be sent to the main office; it directs the call to the main office.
2. The branch office sends the call to 53456 to the main office.
3. The main office determines that this is to another main office. The system transmits 53456 to the NRS. The NRS checks its provisioning, and determines that this call goes to main office B.

In the following diagram, the second half of the call is shown (destination side of the call).

Figure 46
Call flow for Scenario 3- calls to remote branch office (destination side)



1. Main office B determines that this is to a local branch office with prefix 225. The system transmits 225-53456 to the NRS. The NRS checks its provisioning, and determines that this call goes to branch office B1.
2. The main office sends the call to 225-53456 to the branch office. The branch office deletes the prefix and LSC, and rings set 3456.

Emergency Services configuration

Contents

This section contains the following topics:

Overview

Support for access to emergency services by branch users in Normal Mode is configured at the main office.

The key difference between the main office user and the branch user is the route selected for the emergency call. An emergency call must be handed off to the PSTN over a trunk at the central office that is geographically closest to the caller—this means that there is normally an emergency trunk in the main office, and one in each of the branch offices. An emergency call originating from an SRG IP Phone must route from the main office Call Server to the SRG so that the call can be sent on the SRG PSTN Trunks.

In Normal Mode, an IP Phone must have a Virtual Trunk available and configured between the main office and branch office in order to complete an emergency services call.

ATTENTION
Do not route ESA calls to a node that has no direct ESA trunks.

Nortel recommends using the Emergency Services Access (ESA) feature. This is the preferred method in North America, the Caribbean and Latin America (CALA), and in those countries that are members of the European Union (EU). ESA provides specific features and capabilities required by legislation in these jurisdictions.

The ESA feature provides the following advantages:

- recognizes special Emergency Service Directory Number (ESDN)
- overrides calling restrictions
- provides optional assignment of ESA CLID per DN
- provides optional selection of a special emergency route
- provides optional routing digits (for NRS resolution)
- provides optional assignment of an Emergency Location Identification Number (ELIN)
- provides On Site Notification (OSN) through an external tool, which traps the emergency call event and records an alarm when an emergency calls are placed at the branch office

For more information about ESA, see *Emergency Service Access Fundamentals* (NN43001-613) .

The main office Call Server forwards emergency services calls to the SRG using a virtual trunk.

Emergency Services Access

The Emergency Services Access (ESA) configuration specifies the digit sequence (a DN) that the user dials to start an emergency call, known as the Emergency Services Directory Number (ESDN). There can only be one ESA configuration per customer and thus only one ESDN per customer, which means that all telephones on the same network must be in the same numbering plan.

With all sites using the same ESDN, a conflict occurs in the NRS because the same ESDN may need to route to different gateways. The conflict is resolved by using a routing digit for each site that the main office adds as it routes the call. The suggested routing digit is the ESN home location code of the SRG, or alternately, the Numbering Plan Area (NPA) code of the SRG if there is not more than one Call Server in the NPA. Virtually any unique digit string (maximum 15 digits) can be used as a routing digit, because the call is sent to the NRS as a Private/Special Number (SPN). In the NRS, SPN have their own separate numbering plan.

The Automatic Number Identification (ANI) data sent to the Public Safety Answering Point (PSAP) identifies the location of the caller. In some constituencies, legislation requires one ANI (DID) per fixed area, so the physical location of the emergency can be approximated based on the telephone number delivered to the PSAP. The ESA feature has a comprehensive scheme that can be used to convert an extension into an appropriate DID.

If the branch office is relatively small, it can be easier to use a single ANI number for the branch office. For more information on this command, see *Software Input Output Reference - Maintenance* (NN43001-711) .

Routing Emergency Services Access (ESA) calls

> **ATTENTION**
> Do not route ESA calls to a node that has no direct ESA trunks.

Ideally, route ESA calls directly over Central Office (CO) trunks to the Public Safety Answering Point (PSAP). In those cases where this routing is not possible, do not route ESA calls to nodes that have no direct ESA trunks.

The implications of routing calls to nodes without direct ESA trunks are as follows:

- At the node without the direct ESA trunks, the node cannot route the ESA call directly to the PSAP. Instead, that node must re-route the call to another node. This re-routing is an unnecessary use of resources.
- If the node is a CS 1000E node, the only tandem trunks are IP Peer trunks. There is no way to specify the appropriate rerouting digits (that is, Prepend Digits) to reroute the ESA call to another node with direct ESA trunks.

Therefore, if unable to route ESA calls directly to the PSAP, the next best practice is to route ESA calls to nodes with direct ESA trunks.

Emergency call routing

A Call Server can provide service to IP phones across multiple emergency jurisdictions. This can also occur with traditional non-IP equipment in the form of remote peripheral equipment (for example, Carrier Remote, Fiber Remote).

An emergency call should be handled by the designated means for the phone location (for example, local security desk or local PSAP). The emergency call should be routed to a service at the current location of the phone.

Configuring ESA for the branch office

For ESA, the main office Call Server forwards the call to the branch office for termination. Calls are redirected over a Virtual Trunk using the NRS. The NRS routes the calls using a special number, referred to in this section as the ESA Special Number.

ESA must be configured and tested on the main office Call Server and the SRG to differentiate between emergency calls originating from IP Phones at each location and calls originating on trunks, which refers to the forwarded emergency call that the SRG receives from the main office for an IP Phone in Normal Mode.

Use the following steps to configure ESA for emergency access at each location:

At the main office:

Step	Action
1	Determine the dialing plan (for example, numbering plan) for ESA calls.
2	Configure the main office emergency trunk (CAMA or PRI). For EMEA, the following trunks are supported: • BRIE (Basic Rate Interface–ETSI based) • PRI (Primary Rate Interface per EURO ISDN) • QSIG on PRI • DPNSS • IP tandem trunks on ISDN
3	Configure the Virtual Trunk at the main office.
4	Configure ESN at the main office.
5	Configure ESA at the main office.
6	Configure the SRG zone on the main office. Configure a zone for each branch office which is used in conjunction with ESA parameters to route an emergency call to the SRG.
7	Test ESDN using a main office telephone.
8	Configure the SRG emergency trunk (CAMA or PRI).
9	Configure the Virtual Trunk at the MG 1000B.
10	Configure ESN at the branch office.

11 Configure ESA at the branch office.

12 Configure the branch office zone on the branch office.

The branch office zone is required for bandwidth management but does not require ESA parameters.

13 Configure the ESN SPN on the branch office.

14 Configure the NRS for the ESA Special Number used.

15 Test ESDN using an analog (500/2500-type) telephone located at the branch office.

16 Test ESDN using an SRG IP Phone in Normal Mode and in Local Mode.

--End--

At the SRG, or branch office:

Step	Action
1	Configure the SRG emergency trunk (CAMA or PRI).
2	Configure the Virtual Trunk at the MG 1000B.
3	Configure ESN at the branch office.
4	Configure ESA at the branch office.
5	Configure the branch office zone on the branch office. The branch office zone is required for bandwidth management but does not require ESA parameters.
6	Configure the ESN SPN on the branch office.
7	Configure the NRS for the ESA Special Number used.
8	Test ESDN using an analog (500/2500-type) telephone located at the branch office.
9	Test ESDN using an SRG IP Phone in Normal Mode and in Local Mode.

--End--

Reregistering to minimally configured branch office

A branch user in Local Mode but who is not physically at the branch can get incorrect emergency service handling.

If the SRG is not provisioned with knowledge of all the ERL in the enterprise, one of two scenarios occurs when an IP Phone reregisters to the branch (either by VO ESA redirection or by fallback to Local Mode):

- If the local TN is provisioned as Manual Update, then the phone inherits the static location data. The static location data probably indicates basic ESA processing (per LD 24) if this is a small branch.
- If the local TN is provisioned as Auto Update, then cached location data in the phone is rejected if undefined locally, and unknown location values (ERL = 0, ECL = 0, LocDesc = Unknown) are assigned. Unknown location indicates default (basic) emergency processing (per LD 24), which is acceptable for a small branch. A system message is also generated to indicate that the phone location data was actually unknown and defaults were used, but emergency calls should be handled correctly.

Minimally configured branches (without LIS support) can be configured as manual update.

Routing configuration for ESA calls on SRG 50

Use the following steps to configure routing for ESA calls for the SRG 50:

Procedure 4
Configuring routing for ESA calls

Step	Action
1	Build a destination code corresponding to the ESA SPN for the branch office.
2	Configure the destination code to absorb the leading digits for the SPN, leaving just the ESDN.
3	Configure the destination code to use a public route to the PSTN trunks.
4	Ensure the Remote access package (00 to 15 under Call Security) assigned to the VoIP trunks has the appropriate Line Pool Access/Bloc for PRI.
5	Ensure there is a Public Prefix of 911 with a length of 3 to match to outgoing digits. This eliminates any delay. As soon as the 3 digits are collected, the call is sent.

--End--

Determining the dialing plan for ESA calls

In many jurisdictions of the United States and Canada, the emergency number must be “911”. The call processor cannot have a DN that conflicts with these digits, but since “9” is often used for NARS AC2 (the local call Access Code), this is not usually a problem.

ESA for international deployment must support the standard emergency number 112 and any emergency numbers in use prior to the EU directive.

In general, ESA calls should leave the network through a trunk at the branch office where the originating telephone is located. To enable this, it is necessary for telephones at each branch office to supply a unique identifying prefix to the NRS when the ESA calls are being routed so that the NRS can select a distinct route for each branch office. This prefix can be configured with the zone data for the SRG telephones. The provisioning of this prefix is an enhancement for branch office.

While a variety of numbering schemes are available, Nortel recommends that customers use 0 + the ESN location code of the SRG + ESDN, where ESDN is:

- for North America and CALA—911
- for members of the European Union—112 and any other emergency numbers in use prior to the EU directive

This number, referred to here as the ESA Special Number, is configured as a special number (SPN) in the NRS so that the Virtual Trunk routes the call to the branch office.

Use Element Manager or the Command Line Interface for the following procedure. See *IP Peer Networking Installation and Commissioning* (NN43001-313) for details.

Procedure 5
Configuring the main office

Step	Action
1	Configure the main office emergency trunk (CAMA or PRI). Configure either analog CAMA or digital PRI to correctly signal the call identification. ESA overrides all restrictions. Configure the trunk with restrictions so that other features cannot access the trunk.
2	Configure the Virtual Trunk using the procedure from *IP Peer Networking Installation and Commissioning* (NN43001-313) .

The Virtual Trunk must be configured to enable emergency calls originating from SRG IP Phones registered at the main office to reach the branch office.

3 Configure ESN.

ESA uses a route number rather than ESN route list index. However, ESN is required at the branch office.

4 Configure Emergency Services Access (ESA) in LD 24.

Table 20
LD 24 Configure Emergency Services Access

Prompt	Response	Description
REQ:	NEW CHG	Add new data, or change existing data.
TYPE:	ESA	Emergency Services Access data block
CUST	xx	Customer number as defined in LD 15
ESDN	xxxx	Emergency Services DN (for example, 911). Up to four digits are accepted.
ESRT		ESA route number
	0-511	Range for Large Systems
	0-127	Range for MG 1000B
DDGT	x...x	Directing Digits (for CAMA Trunks) (for example, 1, 11, or 911). Up to four digits are accepted.
DFCL	x...x	Default ESA Calling Number. The input must be the following lengths: • On a system that is not FNP equipped, 8 or 11 digits are accepted if the first digit of the input is ’1’; otherwise the input must be 7 or 10 digits. • On a system that is FNP equipped, up to 16 digits are allowed.
OSDN	x...x	On-Site Notification station DN. The input must be a valid single appearance internal DN.

You configure OSDN to alert the local security personnel about an emergency call in progress. Leave the ESA route number blank to make test calls without using any trunk resources. If the route number has been configured, remove it by entering “x” at the prompt. Nortel recommends that the system administrator arrange a test call with the Public Services Access Point (PSAP).

5 Test ESDN using a main office telephone to confirm that main office calls exit the main office trunks. The ESA Configuration Audit feature provides CLID Verification (CLIDVER) reports that determine how an emergency call is routed, without actually routing the call. Use LD 20 to generate a CLIDVER report.

Table 21
LD 20 Generate a CLIDVER report

Prompt	Response	Description
REQ:	PRT	Print
TYPE	CLIDVER	CLID Verification
SORTBY	(DN) TN	The output/report is sorted based on this flag. If the response is DN, the overlay prompts the user to enter the DN, and the output is sorted by the DN. If the response is TN, the overlay prompts the user to enter the TN and the output is sorted by the TN.
ESA_ONLY	(YES) NO	Flag used to decide if the report should contain information for ESA call type only or for all call types. If the ESA package is restricted, this input prompt does not appear. The report contains non-ESA data only.
SHORT	(YES) NO	Flag used to decide if the output report should be a Short report of a Detail report.
TN		Terminal Number
	lscu	Format for Large System, where l = loop, s = shelf, c = card, and u = unit
CUST	xx	Customer number as defined in LD 15.
DN	x...x	Directory Number. If no value is entered, the report includes all supported Directory Numbers.
DATE	dd mmm yyy	Date

PAGE	(NO) YES	Data printed on a per page basis.
DES	d...d	Designator

For more information about CLIDVER reports, see *Emergency Service Access Fundamentals* (NN43001-613) .

6 Configure the branch office zone on the main office.

a Configure the branch office zone ESA dialing information in LD 117.

Table 22
LD 117 Configure branch office zone ESA route

Command	Description
CHG ZESA <Zone><ESA Route #><AC><ESA Prefix><ESA Locator>	
	Defines the ESA parameters for the branch office zone, where: • Zone = Zone number for the branch office. • ESA Route # = Virtual Trunk route to SRG. • AC = Access Code to add to dialed digits. If no AC is required, enter AC0 in place of AC1 or AC2. • ESA Prefix = Digit string added to start of ESDN. This is a unique prefix in the NRS. Nortel recommends that users use 0 + ESN location code of the branch office node. An example for location code 725 would be: 0725. • ESA Locator = Direct Inward Dial telephone number sent as part of ANI for use by the PSAP to locate the source of the call.

b Enable the branch office zone ESA in LD 117.

ENL ZBR <Zone> ESA

7 Configure the ESA Special Number at the main office.

Configure the ESA Special Number in the NRS. Using NRS, configure the ESA Special Number defined for the branch office zone. See *IP Peer Networking Installation and Commissioning* (NN43001-313) .

Nortel recommends that customers use “0” + the ESN Location code + ESDN. An example for location code 725 would be 0725911. The zero is recommended to prevent a collision in the ESN data with the HLOC entry.

8 Do the following:

a In LD 86, configure Emergency Service Access Digit Manipulation for AC + ESDN dialing to allow recognition of the ESDN even if AC1 or AC2 is used.

Table 23
LD 86

Prompt	Response	Description
REQ	NEW CHG	Add, Change
CUST	xx	Customer number as defined in LD 15
FEAT	DGT	Digit Manipulation
...	...	...
DMI	1- 999	Digit Manipulation Table numbers. **ATTENTION** Do not use Digit Manipulation Table 0, as it results in the incorrect call termination treatment.

b Configure the system to trap the ESDN within the AC1 and AC2 translation tables to reprocess the ESDN locally.

Table 24
LD 86 Configure the system to trap the ESDN and reprocess it locally

Prompt	Response	Description
REQ	NEW CHG	Add, Change
CUST	xx	Customer number as defined in LD 15
FEAT	RLB	Route List Block
...	...	...
RLI	xxx	Route List Index to be accessed
ENTR	xxx	Entry number for NARS/BARS Route List
LTER	YES	Local Termination entry. This allows the AC + ESDN call to be recognized as an Emergency Services Access call.
DMI	1 - 999	Digit Manipulation Table. Use the table configured in LD 86. This allows the digits after the AC to remain in the call register as a called number. **ATTENTION** Do not use Digit Manipulation Table 0, as it results in the incorrect call termination treatment.

c Configure Emergency Service Access call recognition for AC + ESDN dialing in LD 90.

Table 25
LD 90

Prompt	Response	Description
REQ	NEW CHG	Add, Change
CUST	xx	Customer number as defined in LD 15
FEAT	NET	Network translation tables
TRAN	aaa	
TYPE	SPN	
- SPN	911	AC + ESDN is recognized as an Emergency Service Access call.
	112	Use the number configured for ESDN.
- RLI	xxx	Route List Index. Use Route List Index configured in LD 86.

--End--

Procedure 6
Configuring the branch office zone

Step	Action
1	Configure the branch office zone on the branch office. In the branch office, only the zone number and bandwidth/codec selection is configured. Use the same zone number between the branch office and main office. The main office configuration (Procedure 5 “Configuring the main office” (page 105), step 6) provides the branch office zone characteristics (local time, local dialing, and ESA).
2	Configure the routing tables on the SRG. The SRG must recognize the incoming digits on the Virtual Trunk and remove all but the ESDN. The call is routed to a local termination.

--End--

Emergency Services for Virtual Office

The E911 Virtual Office feature allows Virtual Office users, whether they are logged in or logged out of Virtual Office to place an emergency (E911) call to the correct Public Safety Answering Point (PSAP) for their geographical location.

The use of the terms Normal Mode and Local Mode apply to SRG branch user only.

Emergency Services while logged in to Virtual Office

The E911 Virtual Office feature recognizes when a user dials an ESDN and it forces the Virtual Office IP Phone to log out of Normal Mode (into Local Mode) in order to place the emergency call directly from the branch office location to the PSAP.

Emergency Services while logged out of Virtual Office

If 911 is dialed while logged out of Virtual Office the LTPS redirects the 911 call to the local 911 service (PSAP), not the remote Call Server 911 service. The Call Server is provisioned with Emergency Services Access Terminal Numbers (ESTN). The ESTN is used to register the IP Phone with the Call Server. The logged out IP Phone can make ESA calls only.

For more information on emergency services for Virtual Office, see *Emergency Service Access Fundamentals* (NN43001-613) and *Branch Office Installation and Commissioning* (NN43001-314) .

On-Site Notification

The ESA On-Site Notification (OSN) function notifies local security personnel when an emergency call occurs. When an emergency call is placed at the branch office, an external tool traps the notification and records an alarm. This applies to IP Phones that the main office returns in local mode when an emergency call is made, as well as locally connected analog (500/2500-type) telephones.

A LAN port must be enabled on the SRG to support the external tool. For more information, see *SRG 50 Configuration Guide* (NN40140-500) .

Configuring the NRS for ESA SPN

The NRS must be configured for the ESA Special Number (SPN). The NRS uses the ESA SPN to route the emergency call from the main office to the branch office.

Nortel recommends that a consistent pattern be followed for all ESA calls. For example, use 0 + ESN Location code of the branch office node + the ESDN. An example for location code 725 would be: 0725911. The zero is recommended to prevent a collision in the ESN data with the HLOC entry.

For more information, see *IP Peer Networking Installation and Commissioning* (NN43001-313) .

Testing the ESDN number

Use Procedure 7 "Testing ESDN using an SRG telephone" (page 112) to test the ESDN number from any telephone in the branch office.

Procedure 7
Testing ESDN using an SRG telephone

Step	Action

1 For IP Phones:

a Dial the ESDN on an SRG IP Phone in Local Mode.

The calls must go out on the emergency trunk(s) in the branch office.

b Dial the ESDN on an SRG IP Phone in Normal Mode.

The calls must tandem over the Virtual Trunk to the branch office and go out on the emergency trunk(s) in the branch office. The following configuration problems can occur:

- The call can receive overflow tones. Use LD 96 to view the digits sent to the Virtual Trunk (ENL MSGO dch#).
- If the digits look correct on the main office, the NRS might not be properly configured. If the NRS rejects the call, a diagnostic message is displayed on the NRS console.
- If the call makes it to the correct branch office (check that it is not going to the wrong node if the NRS is configured incorrectly), the branch office is probably rejecting it because it does not know the digit string.

2 For analog (500/2500-type) telephones, dial the ESDN on an SRG analog (500/2500-type) telephone.

The calls must go out on the emergency trunk(s) in the branch office.

--End--

Configuring ESA using Element Manager

To configure Emergency Services Access in Element Manager, see *Element Manager System Reference - Administration* (NN43001-632) .

Emergency Service using Special Numbers (SPN)

Determining the dialing plan for emergency access calls is critical.

In many jurisdictions, the emergency number is a fixed number (for example, 112 or 999). The main office Call Server or SRG cannot have a DN that conflicts with these digits.

Access to Emergency Service using SPN should be configured in the following circumstances:

- When the Emergency Service number at the branch office is different from that at the main office.
- When there is more than one number used for accessing Emergency Service; for example, when there are different numbers for Police, Fire, and Ambulance services.
- In markets where the ESA feature is not available (outside of North America, CALA, and EMEA).

To configure Emergency Service using SPN, follow the process outlined in "Dialing Plan configuration" (page 39). If SRG PSTN access is correctly configured, Emergency Service from the branch office will already be present.

Branch office access to Emergency Service using SPN must be configured and tested the main office Call Server and the SRG to differentiate between emergency calls originating from IP Phones at each location and emergency calls originating on trunks.

The special handling provided by ESA is not available in this scenario, such as OSN and zone-based routing.

For information on emergency services for Virtual Office, see *Emergency Service Access Fundamentals* (NN43001-613) .

Enhanced UNIStim Firmware Download

Contents

This section contains the following topics:

Description

This section applies to the main office and the following IP Phones:

- IP Phone 2001
- IP Phone 2002
- IP Phone 2004
- IP Phone 2007
- IP Audio Conference Phone 2033
- IP Phone 1110
- IP Phone 1120E
- IP Phone 1140E
- IP Phone 1150E
- IP Phone 1210
- IP Phone 1220
- IP Phone 1230

This section does not apply to the IP Softphone 2050 and WLAN 2210/2211/2212/6120/6140.

The redirected IP Phones at the SRG 50 are under the control of the main office Call Server for the majority of the deployment (Normal Mode). Users of the SRG IP Phones receive the features, key layout, and tones of the main office Call Server. Therefore, the version of the IP Phone firmware must align with the requirements of the CS 1000. When an IP

Phone requires firmware upgrade, the CS 1000 uses the **`umsUpgradeAll`** command, or variant, to redirect the IP Phone back to the SRG 50 for upgrading.

For CS 1000 Release 4.5 and later, if the required firmware file does not exist on the SRG 50, or the version of the file is incorrect, the SRG 50 initiates an FTP session to the TPS for the IP Phone to retrieve the required file. The SRG 50 upgrades the IP Phone and redirects the IP Phone back to the CS 1000.

For SRG 50 Release 2.0 and later, if the required firmware does not exist on the SRG 50, or the version is incorrect, the SRG 50 initiates an FTP session to the TPS for the IP Phone to retrieve the required file. The SRG 50 upgrades the IP Phone and redirects the IP Phone back to the CS 1000.

For SRG 50 Release 1.0, the SRG 50 must be patched to the proper firmware level with patch number BCM50.90. For CS 1000 Release 4.0, ensure MPLR21148 is installed on the Signaling Server. Firmware download does not occur when IP Phones register to the TPS by a Virtual Office Login or branch office redirection to the main office. Instead, SRG IP Phones are redirected back to the SRG TPS for firmware files upgrade. This redirection occurs only if the umsUpgradeAll command is issued from the main office TPS, and the current firmware files are missing.

For CS 1000 Release 4.0, ensure MPLR21148 is installed on the Signaling Server.

Firmware download does not occur when IP Phones register to the TPS by a Virtual Office Login or branch office redirection to the main office. Instead, SRG IP Phones are redirected back to the SRG TPS for firmware files upgrade. This redirection occurs only if the umsUpgradeAll command is issued from the main office TPS, and the current firmware files are missing.

If an IP Phone is in use when the umsUpgradeAll command is issued, the call is not interrupted. Its firmware version is checked against the main office TPS firmware policy, and if there is no match, the IP Phone is flagged, then redirected to the MG 1000B TPS when the call is completed. The umsUpgradeAll command has no immediate impact on IP Phones that are logged in or out by Virtual Office. However, the firmware files may be upgraded, if required, when the Virtual Office session is terminated.

For information on Enhanced UNIStim Firmware, see *Signaling Server IP Line Applications Fundamentals* (NN43001-125) .

Firmware upgrade

Use Procedure 8 “Upgrading firmware” (page 117) to upgrade the firmware. For information about upgrading IP Phone firmware, see *Signaling Server IP Line Applications Fundamentals* (NN43001-125) .

Procedure 8
Upgrading firmware

Step	Action
1	At the Main office, upgrade IP Phone firmware on the Signaling Server. For instructions, see *Signaling Server IP Line Applications Fundamentals* (NN43001-125) .
2	Issue the CLI command **`umsUpgradeAll`** at the main office. IP Phones at the Main office and branch office are upgraded as necessary.

--End--

Appendix
Media Redirection Scenarios

In addition to basic call scenarios, Network Bandwidth Management also supports the following media redirection scenarios:

- Scenario 1: Codec switches correctly during media redirection. See Table 26 "Codec switches correctly during media redirection" (page 120).
- Scenario 2: Call transfer works correctly with IP Phones:
  - Scenario 2.1: Call Transfer from an SRG IP Phone in Normal Mode to main office IP Phone. See Table 27 "Call transfer from SRG IP Phone in Normal Mode to main office IP Phone" (page 120).
  - Scenario 2.2: Call Transfer from main office IP Phone to an SRG IP Phone in Normal Mode. See Table 28 "Call transfer from main office IP Phone to SRG IP Phone in Normal Mode" (page 120).
- Scenario 3: Conference Call works correctly with a branch office:
  - Scenario 3.1: Conference call between branch office and main office, initiated by an SRG IP Phone in Normal Mode. See Table 29 "Conference call between branch office and main office, initiated by SRG IP Phone in Normal Mode" (page 121).
  - Scenario 3.2: Conference call between main office and branch office, initiated by main office IP Phone. See Table 30 "Conference call between main office and branch office, initiated by main office IP Phone" (page 122).

The zone table is examined using the commands in LD 117. See *Software Input Output Reference - Maintenance* (NN43001-711) for more information on these commands.

In these scenarios, consult the zone table at the main office for accurate bandwidth usage information.

Table 26
Codec switches correctly during media redirection

Event	Result
1 An incoming Direct Inward Dial (DID) call to branch office uses IP Peer to reach the symposium controller Control Directory Number (CDN) in the main office.	The external caller hears music and announcements with a G.729 codec. Bandwidth usage in the main office indicates the call is an interzone call. The external caller is connected to an Automatic Call Distribution (ACD) agent with a G.711 codec. Bandwidth usage in the main office indicates the call is an intrazone call. The ACD agent is an SRG IP Phone registered to the main office.
2 The call is released.	The zone table indicates the bandwidth usage for the call is removed correctly on the main office Call Server and in the branch office.

Table 27
Call transfer from SRG IP Phone in Normal Mode to main office IP Phone

Event	Result
1 An SRG TDM telephone calls an IP Phone registered to the main office.	A speech path is established between the SRG TDM telephone and the IP Phone registered to the main office. The zone table indicates intrazone bandwidth usage.
2 The SRG IP Phone registered to the main office initiates a call transfer to a main office IP Phone.	The SRG TDM telephone is put on hold. A speech path is established between the SRG IP Phone registered to the main office and the main office IP Phone. The zone table indicates interzone bandwidth usage.
3 The **Call Transfer** key on the SRG IP Phone registered to the main office is pressed to complete the call transfer.	A speech path is established between the SRG TDM telephone and the main office IP Phone. The zone table indicates interzone bandwidth usage.
4 The call is released.	The zone table indicates bandwidth usage for the call is unreserved correctly.

Table 28
Call transfer from main office IP Phone to SRG IP Phone in Normal Mode

Event	Result
1 An SRG TDM telephone calls a main office IP Phone.	A speech path is established between the SRG TDM telephone and the main office IP Phone. The zone table indicates interzone bandwidth usage.

Table 28
Call transfer from main office IP Phone to SRG IP Phone in Normal Mode (cont'd.)

Event	Result
2 The main office IP Phone initiates a call transfer to an SRG IP Phone registered to the main office.	The SRG TDM telephone is put on hold. A speech path is established between the main office IP Phone and the SRG IP Phone registered to the main office. The zone table indicates interzone bandwidth usage.
3 The **Call Transfer** key on the main office IP Phone is pressed to complete the call transfer.	A speech path is established between the IP Phone registered to the main office and the SRG TDM telephone. The zone table indicates intrazone bandwidth usage.
4 The call is released.	The zone table indicates bandwidth usage for the call is unreserved correctly.

Table 29
Conference call between branch office and main office, initiated by SRG IP Phone in Normal Mode

Event	Result
1 An SRG TDM telephone calls an SRG IP Phone registered to the main office.	A speech path is established between the SRG TDM telephone and the SRG IP Phone registered to the main office. The zone table indicates intrazone bandwidth usage.
2 The SRG IP Phone registered to the main office initiates a conference call to a main office IP Phone.	The SRG TDM telephone is put on hold. A speech path is established between the SRG IP Phone registered to the main office and the main office IP Phone. The zone table indicates interzone bandwidth usage.
3 The Conference key on the SRG IP Phone registered to the main office is pressed to complete the conference call.	Speech paths are established among the SRG TDM telephone, the SRG IP Phone registered to the main office, and the main office IP Phone. The zone table indicates interzone and intrazone bandwidth usage.
4 The SRG TDM telephone releases the call.	A speech path is established between the main office IP Phone and the SRG IP Phone registered to the main office. The zone table indicates interzone bandwidth usage.
5 The call is released.	The zone table indicates bandwidth usage for the call is unreserved correctly.

Table 30
Conference call between main office and branch office, initiated by main office IP Phone

Event		Result
1	An SRG TDM telephone calls a main office IP Phone.	A speech path is established between the SRG TDM telephone and the main office IP Phone. The zone table indicates interzone bandwidth usage.
2	The main office IP Phone initiates a conference call to an SRG IP Phone registered to the main office.	The SRG TDM telephone is put on hold. A speech path is established between the main office IP Phone and the SRG IP Phone registered to the main office. The zone table indicates interzone bandwidth usage.
3	The **Conference** key on the main office IP Phone is pressed to complete the conference call.	Speech paths are established among the SRG TDM telephone, the SRG IP Phone registered to the main office, and the main office IP Phone. The zone table indicates interzone and intrazone bandwidth usage.
4	The SRG TDM telephone releases the call.	A speech path is established between the SRG IP Phone registered to the main office and the main office IP Phone. The zone table indicates interzone bandwidth usage.
5	The call is released.	The zone table indicates bandwidth usage for the call is unreserved correctly.

List of terms

Branch office

An SRG that is remote from the main office. The SRG provides telephony services using the main office servers (for Normal Mode) or local system services when the SRG loses IP communication with the main office (Local Mode).

CDP

Coordinated Dialing Plan. Under the recommended Coordinated Dialing Plan, the Branch User ID can be an extension (for example, 4567). For more information about CDP, see *Dialing Plans Reference* (NN43001-283) .

dialing plan

Each system uses a specific numbering configuration (dialing plan) that determines how calls will be handled over a private or public network.

DSP

Digital Signal Processing, which refers to manipulating analog information, such as sound or photographs that have been converted into a digital form. DSP also implies the use of a data compression technique.

When used as a noun, DSP stands for Digital Signaling Processor, a special type of coprocessor designed for performing the mathematics involved in DSP. Most DSP are programmable, which means that they can be used for manipulating different types of information, including sound, images, and video.

ESA

Emergency Services Access is a feature that places a customer in compliance with federal legislation that requires the Private 911 type of functionality provided by ESA. Please note, however, that the ESA feature is also generally useful for users who are not subject to legislation, and is broad enough to be used in different countries. For example, it will be appreciated by any customer who wants to route emergency calls in a special manner, or who wants to be notified when a telephone user makes an emergency call. It would also appeal to a customer who wishes to have ESA calls answered on-site,

on the business premises, rather than being forwarded to the Public Services Answering Point (PSAP). See *Emergency Service Access Fundamentals* (NN43001-613) for complete information.

Gatekeeper

The Gatekeeper is a separate application on an IP network that directs IP traffic to all the systems on the network. Parameters for both the main office and SRG must be assigned to all gatekeepers active on the network. If the Gatekeeper is down, the SRG attempts to connect to the Alternate Gatekeeper, if there is one. If the Alternate Gatekeeper is down as well, or there is no Alternate Gatekeeper, the SRG IP Phones remain registered with the main office but calls cannot be sent to the SRG.

gateway

In networking, a combination of hardware and software that links two different types of networks. Gateways between e-mail systems, for example, enable users on different e-mail systems to exchange messages.

H.323

A standard approved by the International Telecommunication Union (ITU) that defines how audiovisual conferences data is transmitted across networks. In theory, H.323 enables users to participate in the same conference even though they are using different video conferencing applications. Although most video conferencing vendors have announced that their products conform to H.323, it is too early to say whether such adherence actually results in interoperability.

IP

Abbreviation of Internet Protocol, pronounced as two separate letters. IP specifies the format of packets, also called datagrams, and the addressing scheme. Most networks combine IP with a higher-level protocol called Transport Control Protocol (TCP), which establishes a virtual connection between a destination and a source.

IP by itself is something like the postal system. It enables you to address a package and drop it in the system, but there's no direct link between you and the recipient. TCP/IP, on the other hand, establishes a connection between two hosts so that they can send messages back and forth for a period of time.

LAN

Local Area Network.

Local Mode

The SRG is in Local Mode when:

- The IP Phones are first installed and not yet reregistered with the main office
- The SRG cannot communicate with the main office and the IP Phones are reregistered with the SRG
- A user deliberately puts the IP Phone in the Test Local Mode condition.

Main office

The CS 1000 system that has been programmed to accept redirection of the SRG IP Phones and provide call service for the SRG in Normal Mode.

NCS

Network Connection Service. It provides a TPS interface to the NRS, allowing the TPS to query the NRS using the UNIStim protocol. It is required to support the main office, branch office, Virtual Office, and Geographic Redundancy features.

Normal Mode

The SRG is in Normal Mode when the IP Phones on the SRG are correctly redirected to the main office Call Server.

NRS

Network Routing Service. The software application where all systems in the network are registered. The NRS consists of the H.323 Gatekeeper and the Network Connection Service (NCS).

PSTN

Public Switched Telephone Network. The international telephone system based on copper wires carrying analog voice data. This is in contrast to newer telephone networks based on digital technologies.

Telephone service carried by the PSTN is often called plain old telephone service (POTS).

QoS

Quality of Service, a networking term that specifies a guaranteed throughput level. One of the biggest advantages of ATM over competing technologies, such as Frame Relay and Fast Ethernet, is that ATM supports QoS levels. This enables ATM providers to guarantee to their customers that end-to-end latency does not exceed a specified level.

There are several methods to provide QoS, as follows:

- high bandwidth
- packet classification
- DiffServ
- IP fragmentation
- traffic shaping
- use of the platform’s queuing mechanisms

routing

The process of selecting the correct path for packets transmitted between IP networks by using software-based algorithms. Each packet is processed by the algorithm to determine its destination.

SRG 50

Survivable Remote Gateway 50. This describes the equipment used to create an IP branch office with a CS 1000 system acting as the main office. The base system for SRG is a Business Communication Manager running BCM 3.6 software.

TPS

IP Phone Terminal Proxy Server. This server controls the connection of IP Phones. It resides on the Signaling Server with an emergency backup on the Voice Gateway Media Card.

UDP

Uniform Dialing Plan. Each location within the network is assigned a Location Code, and each telephone has a Directory Number that is unique within the network. Under the UDP, the SRG must include the location code in the Branch User ID (BUID).

VoIP

Voice over IP trunk. This IP pathway between two system IP voice gateways allows the system to exchange telephone calls over the Internet.

WAN

Wide Area Network. A computer network that spans a relatively large geographical area. Typically, a WAN consists of two or more local area networks (LAN).

Computers connected to a wide area network are often connected through public networks, such as the telephone system. They can also be connected through leased lines or satellites. The largest WAN in existence is the Internet.

ZDP

Zone Digit Prefix. This is the number that the main office appends to a local SRG PSTN call dialed from an SRG IP Phone in Normal Mode. This number differentiates the call from a main office PSTN call dialed by the main office telephones. The ZDP routes the call through VoIP trunk to the SRG.

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