Survivable Remote Gateway 50 Configuration Guide

Part No. N0060720 02
June 2006

Task List

Contents

Chapter 1
Getting started

The Survivable Remote Gateway 50 (SRG50) is a new member of the Nortel family of survivable IP telephony branch office solutions that offer business continuity and public switched telephone network (PSTN) failover for voice over IP (VoIP) networks. This family includes the SRG 1.0 and the Media Gateway 1000B in addition to the SRG50. An SRG provides transparent operation and feature parity with a main office call server while in normal operating mode. If connectivity with the call server or wide area network (WAN) is lost, the SRG takes ownership of call control for the local sets automatically, and provides internal communications as well as external connectivity to the PSTN.

The SRG50 supports up to 32 survivable IP users. It is provided as a cost-effective VoIP business continuity solution for small branch offices. The SRG50 is supported on CS 1000 and CS 2000 call servers.

Note: Currently, the SRG50 is a First Customer Application for the CS 2000 and is working through the Nortel Verification Office to achieve full general availability in conjunction with the Centrex IP Client Manager solution.

Information in the *Survivable Remote Gateway 50 Configuration Guide* pertains to Release 1.0 of the SRG50.

Getting started with SRG50 configuration involves reviewing the following material:

Intended audience

This guide is intended for two audiences:

- the individuals responsible for engineering the SRG50 site and installing the BCM50, configuring it for operation as an SRG50, and connecting it to the network
- the individuals responsible for post-installation system administration and maintenance.

The SRG50 site engineer and installer must be familiar with BCM50 hardware and software, and IP telephony and VoIP trunk configuration on the BCM50.

Creating the SRG50

The SRG50 is a software application that leverages the BCM50 platform. It is optimized to provide feature transparency to the main office call server and to act as a survival remote gateway in a branch office environment. An SRG50 is created by applying the SRG application update and enabling the SRG50 application.

The *BCM50 First Time Installation and Configuration Guide* (N0027149) is provided on the SRG50 Release 1.0 Documentation CD that is shipped with your SRG50 system. Instructions in that guide are referenced in the following procedures. Also, the relevant BCM50 default IP addresses, user names, and passwords are excerpted from that guide and provided below for your reference.

Default IP addresses

Port	IP address	Subnet mask
OAM port (see Note)	10.10.11.1	255.255.255.252
BCM50 LAN (no router)	192.168.1.2	255.255.255.0
Note: DHCP is enabled on this port and assigns the following IP address: **10.10.11.2**		

Default user names and passwords

Tool	User ID| User Name	Password
Element Manager	nnadmin	PlsChgMe!
Onbox main web page (http:// *[IP address]*	nnadmin	PlsChgMe!

Applying the SRG application update

To apply the SRG application update

1. Have the SRG50 Release 1.0 Documentation CD and the SRG50 Application Update CD on hand.
2. Insert the SRG50 Release 1.0 Documentation CD into the CD/DVD drive of your client PC.
3. Copy the *BCM50 First Time Installation and Configuration Guide* (N0027149) to the hard drive of the PC.
4. If not done already, download and install the BCM50 Element Manager on your client PC (refer to the Task List in the *BCM50 First Time Installation and Configuration Guide*).
5. Load the SRG50 Application Update CD into your PC.
6. Launch the BCM50 Element Manager.
7. Connect to the BCM50 system (refer to the Task List in the *BCM50 First Time Installation and Configuration Guide*).
8. Navigate to the **Date and Time** panel (**Configuration > System > Date and Time**).
9. On the **Current Date and Time** subpanel, change the **Date and time** as required and select the appropriate time zone from the **Time zone** pull-down menu.
10. Navigate to the Software Updates panel (**Administration > Software Management > Software Updates**).
11. Click **Get New Updates** button.
12. From the **Retrieve from** pull-down menu, select **My Computer**.
13. Browse to the SRG50 Application Update CD and click **Select**.

    An update summary appears.
14. Select the file whose filename begins with **SRGP** and click **Apply**.

The update proceeds. The BCM50 reboots automatically at the end of the application.

Note: There is a delay between successful update of the application and the automatic reboot of the system. Wait for the system to reboot before proceeding.

Applying SRG50 keycodes

Four keycodes are available for SRG50. The initial keycode enables the SRG50 application, and includes:

- eight survivable IP users
- up to 24 H.323 VoIP trunks
- MCDN/QSIG enabled

- four PSTN trunks (FXO) (not supported in Category 2* countries)
- four analog station interfaces (not supported in Category 2* countries)
- automatic activation of Meridian Customer Defined Network (MCDN) upon conversion to SRG operation
- 12 digital interfaces (for ATA2 use; blocked for digital set use)
- both BCM50 expansion ports, where each expansion port can support one Media Bay Modules (MBM)
  - — digital trunk support in Category 1* and 2* countries
  - — analog trunk and analog station support in Category 2* countries using standard MBMs, or, alternatively, using an ATA2 on the digital interface of the main chassis

    * Contact your Nortel representative for information on Category 1 and Category 2 countries.

Up to three additional keycodes can be applied to increase the number of survivable IP users in increments of eight for a total of 16, 24, or 32 users.

To enable the SRG50 application

After the SRG50 application update is installed and the BCM50 has rebooted, the first SRG50 keycode must be applied to enable the SRG50 application.

1. Apply the SRG50 application update as instructed in the preceding procedure.
2. Locate the Authorization codes.
3. Obtain the keycode(s) for your SRG50 system from Nortel's Keycode Retrieval System at:

   http://www.nortel.com/servsup/krs/

   (For product family, choose Survivable Remote Gateway 50.)
4. If not already open, launch the BCM50 Element Manager.
5. If not already connected, connect to the BCM50 system (refer to the Task List in the *BCM50 First Time Installation and Configuration Guide*).
6. Navigate to the **Keycodes** panel (**Configuration > System > Keycodes**).
7. Click the **Load File** button.
8. Browse to the folder containing the downloaded keycode file(s) for this system.
9. Select the initial keycode file and click **Open**.

   The keycode file is applied and you are prompted to reboot the system.

> **Note:** You must reboot the system.

To reboot the system

1 Navigate to the **Reset** panel (**Administration > Utilities > Reset**).

2 Click the **Reboot BCM50 System** button.

To verify that the SRG50 has been created successfully

1 Apply the SRG50 application update and first keycode as instructed in the preceding procedures.

2 Launch the BCM50 Element Manager.

3 Connect to the BCM50.

4 Navigate to the **System Identification** panel (**Configuration > System > Identification**).

5 Verify that the entry in the **Description** field is **SRG (Telephony Only)**.

6 Navigate to the **Keycodes** panel (**Configuration > System > Keycodes**).

7 In the **Feature licenses** table, verify that the **Status** of the **SRG** keycode is **ACTIVE**.

8 Open the **Resources** folder (**Configuration > Resources**).

9 Verify that there is a **Survivable Remote Gateway** panel.

Warning: Reversability — A Level 1 or Level 2 reset causes the SRG50 to revert to BCM50 functionality. Refer to the *BCM50 Installation & Maintenance Guide* (N0027152) for details.

To increase the number of survivable IP users

After the initial keycode has been applied and the system has been rebooted, the number of survivable IP users can be increased to 16, 24, or 32 by the application of additional keycodes. To apply these keycodes, follow the procedure provided above for applying the initial keycode.

Note: You can apply one, two, or three keycodes in one session but you must reboot the system for the keycodes to take effect. Once additional keycodes have been applied, the **Data** column of the **Feature licenses** table (**Configuration > System > Keycodes**) updates to reflect the new number of users.

Comparison of SRG50 and BCM50 features

The following table compares SRG50 and BCM50 features.

Table 1 Comparison of BCM50 and SRG50

Item	**BCM50**	**SRG50**
MBMs	Refer to the *BCM50 Device Configuration Guide* (N0027146)	Recommended: ASM8+ (8 port Analog Station Module); DTM (Digital Trunk Module - 24 lines on either T1 or E1 or PRI); BRI (4 line BRI S/T Module); GATM4 (Global Analog Trunk MBM - 4 port); GATM8 (Global Analog Trunk MBM - 8 port) Supported for ATA connections: DSM16 (Digital Station Module - 16 ports); DSM32 (Digital Station Module - 32 ports); 4x16 Combo (16 digital ports, 4 analog trunks and 1 analog station) Does not support: DDIM (Digital Drop and Insert Mux)
Digital telephone sets	Yes	No
FCAPS	Yes	Yes, extended to include SRG-specific alarms and keycodes
Network Configuration Manager	Yes	No
Telset Administration	Yes	No
CS 1000 Geographic Redundancy	N/A	Yes
CS 1000 Network Bandwidth Management	N/A	Yes
CS 1000 Adaptive Network Bandwidth Management	N/A	Yes
CS 1000 Alternative Call Routing	N/A	Yes
CS 1000 Emergency Services Access	N/A	Yes
Firmware Download from main office call server	N/A	Yes (CS 1000 Release 4.5 only)
SRG-specific features for interaction with a main office call server, including: Heartbeat detection of WAN recovery; IP telephone redirection to main office in Normal Mode; Local Mode IP telephone interface; H.323 Gateway to PSTN under control of main office call server (CS 1000 only)	N/A	Yes

Devices supported by the SRG50

The SRG50 supports:

- Nortel 2001, 2002, 2004, and 2007 IP telephones
- Nortel 2050, MVC2050 and MVC2050E Softphones (CS 1000 only)
- Nortel 2210 and 2211 wireless LAN (WLAN) handsets
- analog telephones

- analog devices such as fax machines

The SRG50 is positioned primarily to support IP telephones and clients. However, analog devices can be supported using analog station modules (ASM), or by using an analog terminal adapter (ATA2) in conjunction with a digital station module (DSM). The SRG50 does not support digital or ISDN telephones.

SRG50 terminology

The following table identifies SRG50 terms that may be unfamiliar to main office installers. They are provided to facilitate communications between SRG50 and main office personnel. In the table, EM refers to a path on the SRG50 Element Manager where the term appears; the paths are provided for reference and may not represent every appearance of the term.

Term	Description
Port	For telephony configuration (**EM: Configuration > Telephony**), a port is an internal number that identifies a physical termination point for a telephone set or a physical trunk. For the configuration of resources (**EM: Configuration > Resources**) and data services (**EM: Configuration > Data Services**), port is used in the context of the TCP/IP protocol suite.
IP Terminal	IP telephone **EM: Configuration > Resources > Telephony Resources > IP & Application Sets**
Sets	Can refer to actual telephones, or to the directory number (DN) assigned to the port to which a particular telephone is connected. Telephone **EM: Configuration > Resources > Telephony Resources > IP & Application Sets** Mapping DN to Telephone **EM: Configuration > Telephony > Sets** DN **EM: Configuration > Telephony > Lines > Target Lines > Target Lines** table **>** **Control Set** and **Prime Set** columns
Trunks	Trunks refer to external facilities that are connected to the SRG50 and provide incoming and outgoing communication paths. Paths can be physical (examples: loop; PRI; T1) or virtual (VoIP trunks). **EM: Configuration > Resources**
Loop trunk	An analog loop (FXO) that connects to the PSTN: a POTS line.
Lines	A line is the generic term used for all communication paths, both internal and external. **EM: Configuration > Telephony > Lines**
Physical Lines	Physical trunks. **EM: Configuration > Telephony > Lines > Active Physical Lines** (Lines 061 to 124)

Term	Description
VoIP Lines	VoIP trunks. **EM: Configuration** > **Telephony** > **Lines** > **Active VoIP Lines** (Lines 001 to 024)
Target Lines	Target lines are internal, virtual paths between trunks and telephones for incoming calls (only). They provide flexibility in the way trunks and telephones can be associated: target lines can be used to direct an incoming call to one or more telephones, direct one or more trunks to one phone, or direct several trunks (in a line pool) to one or more phones. Target lines are assigned to DNs. A target line triggers ringing voltage to the telephone(s) connected to the port(s) associated with the DN(s) that the target line is assigned to. (For example, if a unique target line is assigned to each DN, only one telephone rings when the DN is called. If several DNs are assigned to one target line, calling any of the DNs ring all of the associated phones.) Target lines are required for auto-answer trunks. Because VoIP lines are set internally to auto-answer, target lines are required for SRG operation. The SRG50 Element Manager provides two methods for assigning target lines to DNs. 1) **EM: Configuration** > **Telephony** > **Sets** > **All DNs** > **All DNs** table > **Details for DN** subpanel > **Line Assignment** tab or 2) **EM: Configuration** > **Telephony** > **Lines** > **Target Lines** > **Target Lines** table > **Details for Line** subpanel > **Assigned DNs** tab The first method provides a convenient way to assign the target line to the DN when the DN record is configured. The second method provides fields that allow incoming digit strings to be mapped to the DN. (Lines 125 to 268) For more information on target lines, refer to the *BCM50 Networking Configuration Guide* (N0027156).
Line pool	Several of the same type of trunk configured as one group: a trunk group.

Coordination with the main office call server

Configuration of the SRG50 branch office requires datafill at both the SRG50 and the main office call server. Main office configuration drives SRG50 configuration, and Nortel recommends that the main office activities be concluded before undertaking SRG50 configuration.

Note: Configuration activities at the SRG50 that are unique to a specific type of main office call server are covered in separate chapters in this guide.

- CS 1000
  Refer to Chapter 3, "CS 1000 considerations," on page 33 for information specific to Nortel Communication Server 1000.
- CS 2000
  Refer to Chapter 4, "CS 2000 considerations," on page 53 for information specific to Nortel Communication Server 2000.

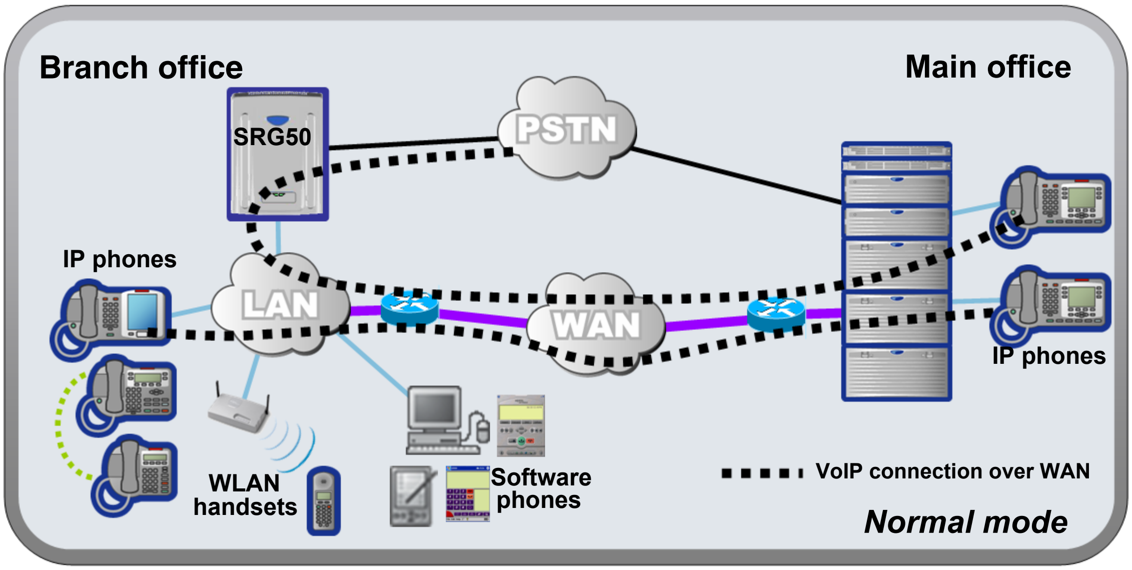
SRG50 operating modes

The SRG50 has two operating modes: normal and local.

Normal mode

In normal mode (Figure 1), the SRG50 is connected to the main office call server over a WAN using VoIP trunks. From the perspective of the main office, the SRG50 is a branch office.

Figure 1 Normal mode



IP telephones connected at the SRG50 are registered with the main office call server and are under main office control. They operate as branch user sets and have access to all telephony services and features that the call server offers to IP telephones connected directly to the main office.

When a branch user set initiates a local PSTN call, the main office sets up the call using the VoIP trunks, which establishes a local media path. Emergency Services Access calls are similarly routed to the SRG50 PSTN. For main office callers, the SRG50 acts as a VoIP-PSTN gateway during normal mode.

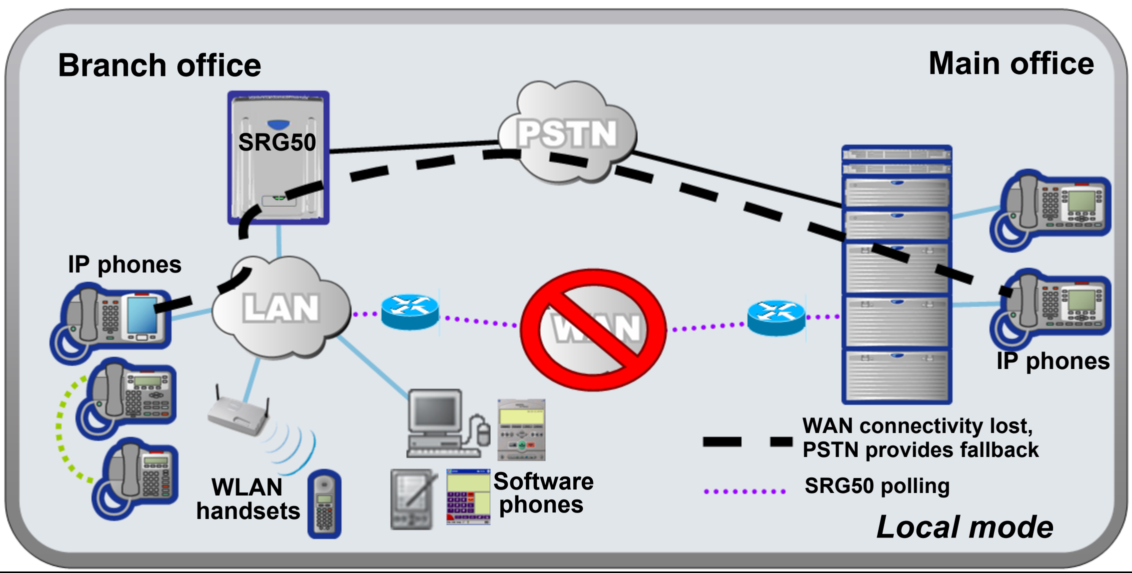
When call forwarding has been configured, incoming PSTN calls to the branch user set are forwarded over VoIP trunks to the main office, which terminates the call at the branch user. Similarly, calls from analog telephones connected to the SRG50 to the branch user set are forwarded to the main office over VoIP trunks, which then terminates the call at the branch user. Calls from the branch user set to the analog telephones at the SRG50 are routed over the VoIP trunks to terminate at the analog telephone. In all these call scenarios, only signaling messages go through the VoIP trunk. The media path is set up directly between the branch user set and the voice gateway at the SRG50. This means that these calls do not use any WAN bandwidth between the main office and the branch office after calls are established.

When a branch user IP telephone calls a main office IP telephone and vice versa, the call is a simple station-to-station call within the main office call server. Since the branch user IP telephone is physically remote from the call server, the media path goes through the WAN connection between the main office and the SRG50, and thus uses WAN bandwidth, as demanded by the codec used in the call.

Local mode

In the event of a WAN failure or the call server at the main office becomes unavailable, the SRG50 reverts to local mode automatically. In local mode, the IP users connected to the SRG50 are under the control of the SRG50. When in local mode, main office call features are not available to users attached to the SRG50. The SRG50 offers a set of basic features for the IP telephones, including access to the local PSTN, dialing emergency service numbers, and calling local extensions. (For a complete list of local mode features, refer to "Features in local mode" on page 73.) Local mode is illustrated in Figure 2.

Figure 2 Local mode



he SRG50 handles all call processing. Calls between two IP telephones at the SRG50 are handled locally as a simple station-to-station call. When an IP telephone initiates a local PSTN call, the SRG50 routes the call to a trunk that is connected to the local PSTN. Incoming DID calls are also handled by the SRG50 and terminated on the appropriate IP telephone set.

In local mode, the IP telephones do not have access to the main office network over the VoIP trunks. If alternate routes are configured, then calls can be made to the main office or other branch offices using the available PSTN trunks.

Several situations, described below, can cause the IP phone to be in local mode.

Initial registration, CS 1000

When the IP telephone is installed, it first registers with the SRG50, and is in local mode. When the SRG configuration at the main office and the SRG50 is complete the IP telephone is redirected to the main office, where it registers as a branch user and changes from local mode to normal mode.

Initial registration, CS 2000

When the IP telephone is installed, it first registers with the CS 2000. When the SRG configuration at the main office and the SRG50 is complete, the IP telephone is redirected to the SRG50 manually, where it registers with the SRG50. The phone is then in local mode.

Failure to register with the main office

When configured as a branch office user set, an IP telephone at the SRG50 automatically attempts to register with the main office when:

- The phone is in local mode because of loss of connectivity with the main office, and the SRG50 is redirecting it back to the main office because connectivity has been reestablished (see “Loss of WAN or VoIP connectivity”below).
- The phone is in local mode because Test Local Mode was invoked and the timer has expired or the Exit button is pressed.
- The phone is in local mode, the main office is a CS 1000, and this is the first time that the phone has been redirected to the main office.

The IP telephone can fail to register with the main office for several reasons. These are detailed in “Probable causes for redirection failure” on page 98.

Loss of WAN or VoIP connectivity

The WAN or VoIP connectivity between the main office and the SRG50 can become unavailable if, for example, router failure occurs, the main office becomes unavailable, a WAN failure occurs, or the VoIP trunks reach capacity. When VoIP connectivity is lost, each IP telephone loses its Reliable UDP (RUDP) connection with the main office terminal proxy server (TPS, CS 1000) or centrex IP client manager (CICM, CS 2000). The IP telephones reboot and reregister at the SRG50, placing them in local mode. If enabled, call forwarding to the main office is automatically cancelled.

The IP telephones remain under the control of the SRG50 until VoIP connectivity is confirmed. When confirmation is received, the IP telephones are automatically redirected to the main office; redirection requires no user intervention. If the telephone is busy at the time that connectivity is reestablished, the SRG50 redirects the phone when it is free.

Test Local Mode

Test Local Mode is a facility that allows the IP telephone to be redirected back to the SRG50 when it is in normal mode. This forces the IP telephone to go into local mode and allows the telephone user or system administrator to test local mode operation without taking down the VoIP trunks to the main office. Implementation of Test Local Mode depends on the main office call server and is covered in the server-specific chapters ("CS 1000 considerations" on page 33 and "CS 2000 considerations" on page 53).

SRG50 installation and configuration summary

The *Survivable Remote Gateway 50 Configuration Guide* provides information specific to configuring a BCM50 as an SRG. Information pertaining to generic BCM50 practices and procedures is provided in the BCM50 documentation suite. This suite is included on the SRG50 CD, and specific documents are referenced in the *Survivable Remote Gateway 50 Configuration Guide* where applicable.

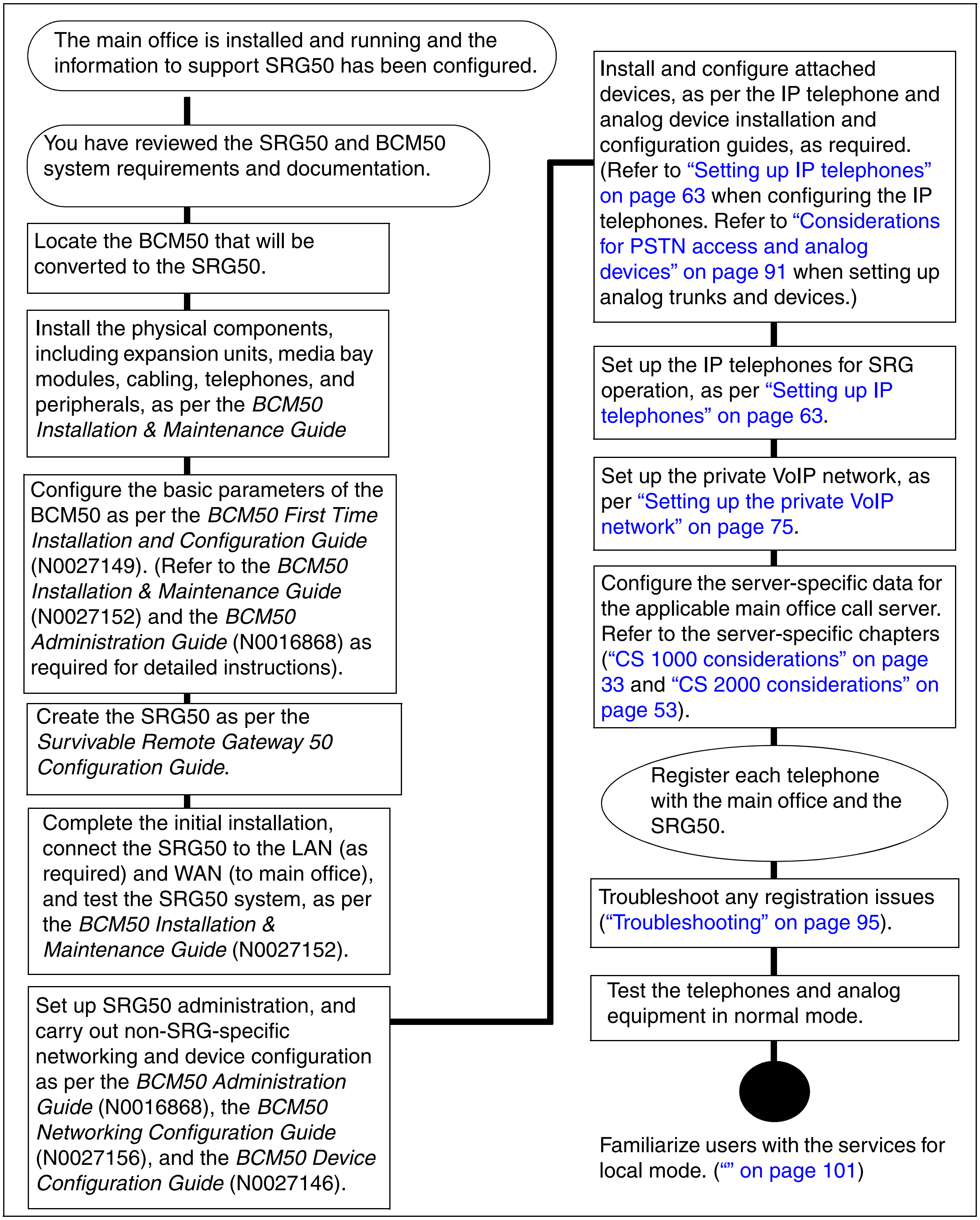
Generally, SRG50 activities follow upon and leverage an installer's general knowledge of BCM50 activities. However, Nortel recommends that the BCM50/SRG50 site engineer and installer familiarize themselves with SRG-specific requirements before starting any installation activities.

Figure 3 on page 21 provides a process map for installing and configuring an SRG50. The procedures in the *Survivable Remote Gateway 50 Configuration Guide* assume that the following activities have been completed:

- The BCM50, including expansion units, media bay modules, cabling, telephones, and peripherals, have been installed.
- BCM50 administration has been set up.
- The basic parameters of the BCM50 have been configured.
- The SRG50 has been connected to the LAN (as required) and WAN (to the main office).
- System functionality has been tested to this point.
- Attached devices have been installed and configured (refer to "Setting up IP telephones" on page 63 when configuring the IP telephones).
- Non-SRG-specific networking and device configuration has been completed (refer to "Setting up the private VoIP network" on page 75 when configuring the network).

Process map for installing and configuring an SRG50

Figure 3 Process map for installing and configuring an SRG50



Acronyms used in this guide

This guide uses the following acronyms:

ATA (or ATA2)	Analog Terminal Adapter
BCM	Business Communications Manager
BO	Branch Office
BUID	Branch User ID
CCR	Custom Call Routing
CDP	Coordinated Dialing Plan
CICM	Centrex IP Client Manager
CODEC	Coder / Decoder
DID	Direct Inward Dialing
DN	Directory Number
EM	Element Manager
FXO	Foreign eXchange Office
IP	Internet Protocol
ISDN	Integrated Services Digital Network
LAN	Local Area Network
MCDN	Meridian Customer Defined Network
MO	Main Office
MOTN	Main Office Telephone Number
MSC	Media Services Card
NCS	Network Connection Server
NPI	Numbering Plan ID
OTM	Optivity Telephony Manager
POTS	Plain Old Telephone Service
PSTN	Public Switched Telephone Network
QoS	Quality of Service
SRG	Survivable Remote Gateway
TPS	(Internet Telephone) Terminal Proxy Server
UDP	Uniform Dialing Plan or User Datagram Protocol
VoIP	Voice over Internet Protocol
VPNI	Virtual Private Network Identifier
WAN	Wide area network

ZACB	Zone Access Code Behavior
ZDP	Zone Digit Prefix

Symbols used in this guide

This guide uses symbols to draw your attention to important information. The following symbols appear in this guide:

Caution: Caution Symbol
Alerts you to conditions where you can damage the equipment.

Danger: Electrical Shock Hazard Symbol
Alerts you to conditions where you can get an electrical shock.

Warning: Warning Symbol
Alerts you to conditions where you can cause the system to fail or work improperly.

Note: Note Symbol
Alerts you to important information.

Tip: Tip Symbol
Alerts you to additional information that can help you perform a task.

Security Note:
Indicates a point of system security where a default should be changed, or where the administrator needs to make a decision about the level of security required for the system.

Text conventions

This guide uses the following text conventions:

angle brackets (< >)	Indicates that you choose the text to enter based on the description inside the brackets. Do not type the brackets when entering the command. Example: If the command syntax is: **`ping`** *`<ip_address>`* you enter: **`ping 192.32.10.12`**
`bold Courier text`	Indicates command names and options and text that you need to enter. Example: Use the **`dinfo`** command. Example: Enter **`show ip`** `{`**`alerts`**`|`**`routes`**`}`.

italic text	Indicates book titles
`plain Courier text`	Indicates command syntax and system output, for example, prompts and system messages. Example: `Set Trap Monitor Filters`
FEATURE **HOLD** **RELEASE**	Indicates that you press the button with the coordinating icon on whichever set you are using.

How to get help

This section explains how to get help for Nortel products and services.

Getting Help from the Nortel Web site

The best source of support for Nortel products is the Nortel Support Web site:

http://www.nortel.com/support

This site enables customers to:

- download software and related tools
- download technical documents, release notes, and product bulletins
- sign up for automatic notification of new software and documentation
- search the Support Web site and Nortel Knowledge Base
- open and manage technical support cases

Getting Help over the phone from a Nortel Solutions Center

If you have a Nortel support contract and cannot find the information you require on the Nortel Support Web site, you can get help over the phone from a Nortel Solutions Center.

In North America, call 1-800-4NORTEL (1-800-466-7865).

Outside North America, go to the Web site below and look up the phone number that applies in your region:

http://www.nortel.com/callus

When you speak to the phone agent, you can reference an Express Routing Code (ERC) to more quickly route your call to the appropriate support specialist. To locate the ERC for your product or service, go to:

http://www.nortel.com/erc

Getting Help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or authorized reseller, you can contact the technical support staff for that distributor or reseller.

Chapter 2
Task summary

The task summary offers a high level, chronological review of the tasks required to configure the SRG50. The paths (**Xxxx > Yyyy > Zzzz**) direct you to the appropriate panels on the SRG50 Element Manager.

Foundation configuration

Foundation configuration refers to configuration that is done as part of BCM50 foundation activities. The items identified here are significant for SRG operation and main office planning and installation.

a Configure the SRG50 IP address, net mask, and gateway.

Configuration > System > IP Subsystem

External Reference: *BCM50 Installation & Maintenance Guide* (N0027152)

b Confirm the number of IP sets and VoIP trunks.

Configuration > Resources > Application Resources
The **Licence** column indicates the number of resources available.

External Reference: *BCM50 Keycode Installation Guide* (N0016865)

c Verify the global telephony settings.

Configuration > Telephony > Global Settings

External Reference: *BCM50 Device Configuration Guide* (N0027146)

d Configure the Start DN (determined by the dialing plan).

Administration > Utilities > Reset > Reset panel **> Cold Reset Telephony Services** button **> Cold Reset Telephony** pop-up window **> Start DN** field

Internal Reference: *"Basic parameters" on page 77*
External Reference: *BCM50 Installation & Maintenance Guide* (N0027152)

e Verify the DN length.

i) For local calls between telephones on the SRG50.

Configuration > Telephony > Dialing Plan > General > Dialing Plan - General panel **> Global Settings** subpanel **> DN length (intercom)** field

ii) For incoming calls from the PSTN
Configuration > Telephony > Dialing Plan > Public Network > Dialing Plan - Public Network panel**> Public Network Settings sub**panel **> Public Received number length** field

iii) For calls coming in from the private network
Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel **> Private Network Settings** subpanel **> Private Received number length** field
and
Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel **> Private Network Settings** subpanel **> Private DN length** field (Private DN length is used for DPNSS applications only. Refer to the *BCM50 Networking Configuration Guide* (N0027156).)

Internal Reference: *"Basic parameters" on page 77*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

f Verify the line pool assignment of VoIP trunks.
In the default configuration, the VoIP trunks are assigned to line pool BlocA. Instructions in the *Survivable Remote Gateway 50 Configuration Guide* assume that the default configuration has been maintained.

Configuration > Telephony > Lines > Active VoIP Lines > Active VoIP Lines table **> Line Type** column

External Reference: *BCM50 Networking Configuration Guide* (N0027156)

g The SRG50 supports four analog loop trunks on the main unit*. Verify the line pool assignment of these trunks.
In the default configuration, these trunks are assigned to line pool A. Instructions in the *Survivable Remote Gateway 50 Configuration Guide* assume that the default configuration has been maintained.

Configuration > Telephony > Lines > Active Physical Lines > Active Physical Lines table **> Line Type** column

External Reference: *BCM50 Networking Configuration Guide* (N0027156)

* Category 1 countries

IP telephone configuration

a Configure the registration password.

Configuration > Resources > Telephony Resources > IP & Application Sets row > **Details for Module: Internal** subpanel > **IP Terminal Global Settings** tab

Internal Reference: "Registration password" on page 63

b Configure the local mode indication.

Configuration > Resources > Telephony Resources > IP & Application Sets row > **Details for Module: Internal** subpanel > **IP Terminal Global Settings** tab

Internal Reference: "Local mode indication" on page 65

c Configure the IP telephone codec and jitter settings.

Configuration > Resources > Telephony Resources > IP & Application Sets row > **Details for Module: Internal** subpanel > **IP Terminal Global Settings** tab

Internal Reference: "IP telephone codec and jitter settings" on page 65

d Configure the telephone (DN) records.

Configuration > Telephony > Sets > All DNs

Internal References:
"Configuring telephone (DN) records" on page 66

e Configure the received numbers.

Configuration > Telephony > Lines > Target Lines

Internal Reference: "Configuring received numbers" on page 69
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

f Decide on the call forwarding option.

Internal Reference: "Call forwarding options" on page 71

g Configure the IP telephones.

Internal Reference: "Configuration settings for redirected phones" on page 72

Dialing plan configuration

a Configure the type of dialing plan (CDP or UDP).

Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel **> Private Network Settings** subpanel

Internal Reference: *"Private dialing plan" on page 78*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

b Enable MCDN TAT.

Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel **> Private Network Settings** subpanel

Internal Reference: *"Meridian Customer Defined Network (MCDN)" on page 79*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

VoIP trunk configuration

a Configure VoIP trunk QoS settings.

Configuration > Resources > Telephony Resources > Modules panel **> IP Trunks** row **> Media Parameters** tab

Internal Reference: *"QoS settings (codec, jitter buffer, and related items)" on page 80*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

b Enable or disable fallback.
Configuration > Resources > Telephony Resources > Modules panel **> IP Trunks** row **> Local Gateway** tab

Internal References:
"Configuring fallback" on page 83
"Call routing: providing access to the SRG50 PSTN" on page 88
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

c Configure gatekeeper settings (CS 1000 only).
Configuration > Resources > Telephony Resources > Modules panel **> IP Trunks** row **> Local Gateway** tab

Internal Reference: *"Gatekeeper routing (CS 1000 only)" on page 84*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

d Assign VoIP trunks to a line pool (if default configuration has not been maintained).

Configuration > Telephony > Lines > Active VoIP Lines

Internal Reference: *"Line pools" on page 86*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

e Assign PSTN trunks to a line pool (if default configuration has not been maintained).

Configuration > Telephony > Lines > Active Physical Lines

Internal Reference: *"Line pools" on page 86*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

f Assign remote access packages to the VoIP trunks.

Internal Reference: *"Call routing: providing access to the SRG50 PSTN" on page 88*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

Call routing configuration

a Decide on the fallback scheme.

Internal Reference: *"Configuring fallback" on page 83*

b Configure the outgoing routes (VoIP and PSTN fallback).

Configuration > Telephony > Dialing Plan > Routing

Internal Reference: *"Call routing: configuring for outgoing calls" on page 86*
External Reference: *BCM50 Networking Configuration Guide*

c Configure access to the SRG50 PSTN (for both local and tandem calls).

Configuration > Telephony > Dialing Plan > Routing

Internal Reference: *"Call routing: providing access to the SRG50 PSTN" on page 88*
External Reference: *BCM50 Networking Configuration Guide* (N0027156)

d Configure for Network Bandwidth Management and Advanced Network Bandwidth Management (CS 1000 only).

Internal Reference: *"Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR" on page 39*
External Reference: *Branch Office: Installation and Configuration* (553-3001-214) and *Data Networking for Voice over IP* (553-3001-160)

e Configure for Alternative Call Routing (CS 1000 only).

Internal Reference: *"Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR" on page 39*
External Reference: *What's New for Communication Server 1000 Release 4.5* (553-3001-015)

Redirection and call forward configuration

a Configure the main office settings.

CS 1000:
Configuration > Resources > Survivable Remote Gateway > S1000 Main Office Settings tab
Internal Reference: "CS 1000 information for the SRG50" on page 43

CS 2000:
Configuration > Resources > Survivable Remote Gateway > CS2000 Main Office Settings tab
Internal Reference: "Configuring IP telephones for redirection" on page 55

b Configure the IP terminal settings.

CS 1000:
Configuration > Resources > Survivable Remote Gateway > S1000 IP Terminal Details tab
Internal Reference: "Configuring IP telephones for redirection" on page 46

CS 2000:
Configuration > Resources > Survivable Remote Gateway > CS2000 IP Terminal Details tab
Internal Reference: "Configuring IP telephones for redirection" on page 55

c Register the IP telephones at the SRG50 (CS 2000 only).

Internal Reference: "Registering the IP telephones at the SRG50: Test Local Mode" on page 61

Chapter 3
CS 1000 considerations

A survivable remote gateway (SRG) extends CS 1000 features from the main office and provides a business continuity solution to one or more remote SRG locations (branch offices). The SRG50 Release 1.0 operates with CS 1000 running Release 4.5 and is backward compatible to Release 3.0 and Release 4.0. The SRG50 does not operate with CSE 1000 Release 1.0 and Succession 1000 2.0 systems. The SRG50 is optimized for a branch office or remote site with a requirement for 5-to-32 survivable IP users.

The *Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50* guide provides information specific to the configuration of an SRG50 on the CS 1000. This guide is included on the SRG50 documentation CD for your reference. Access to other CS 1000 documentation may be required if personnel are not familiar with configuration of branch offices on the CS 1000. The *Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50* guide provides contextual references to these other documents.

This following activities are specific to SRG50 configuration when the main office call server is a Nortel Communication Server 1000 (CS 1000).

- “Cross-reference for CS 1000 and SRG50 terminology” on page 34
- “Normal and local mode overview: CS 1000 details” on page 35
- “Virtual trunk capacity” on page 37
- “Vacant Number Routing” on page 37
- “Bandwidth Management” on page 38
- “Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR” on page 39
- “Configuring for Emergency Services Access (ESA)” on page 42
- “CS 1000 information for the SRG50” on page 43
- “Configuring IP telephones for redirection” on page 46
- “Firmware upgrade” on page 51

Cross-reference for CS 1000 and SRG50 terminology

The following table compares configuration-related terms and contexts of the CS 1000 and the SRG50.

Table 2 Comparison of CS 1000 and SRG50 terms and contexts (Sheet 1 of 2)

Term or Context	CS 1000	SRG50
Dialing plan	on-net / off-net dialing	Private / Public network dialing
Type of number	CDP / UDP / GDP / TNDN	CDP / UDP / no equivalent
Numbers	TN (terminal number)	MOTN (main office terminal number)
	TN = MOTN. That is, the TN from the main office is entered on the SRG50 in the MOTN field (refer to "Configuring IP telephones for redirection" on page 46).	
	BUID (branch user ID) The dialable number of an IP telephone at the SRG50 when it is called from a phone located at the main office or another branch office.	The CS 1000 BUID is entered on the SRG50 (refer to "Datafilling the S1000 IP Terminal Details panel" on page 48) but there is no SRG50 equivalent for BUID.
	DN (directory number) The dialable number of a telephone at the main office when it is called from another phone at the main office.	DN (directory number) The dialable number of a telephone at the SRG50 when it is called from another phone at the SRG50.
	In the case of a CDP dialing plan, it is recommended that the BUID and the SRG50 DN be the same. In the case of a UDP dialing plan, the BUID has the form: <VOIP Trunk Access Code> + <LOC> + <DN>. In this case, it is recommended that the SRG50 DN be the same as <DN>. The dialable number of an IP telephone at the SRG50, when dialed from another phone at the SRG50, remains the same in both normal and local mode if the preceding recommendations are implemented.	
	AC1	VOIP Trunk Access Code (refer to "CS 1000 information for the SRG50" on page 43) Destination code for VoIP trunks (refer to "Call routing: configuring for outgoing calls" on page 86)
	AC1 = VOIP Trunk Access Code = Destination code for VoIP trunks	
Routing	distant steering codes (DSC), trunk steering codes (TSC), local steering codes (LSC)	call routing, destination codes, line pool access codes
	digit manipulation table	dialout digits (routing)
Numbering Plan ID	ISDN/Telephony (E.164), Private, Telephony (E.163), Telex (F.69), Data (X.121), National Standard	Private

Table 2 Comparison of CS 1000 and SRG50 terms and contexts (Sheet 2 of 2)

Term or Context	**CS 1000**	**SRG50**
Access codes (SRG50: Destination codes)	7 = system trunk access 8 = Basic Alternate Route Selection (BARS)/Network Alternate Route Selection (NARS) 9 = public exchange access	7 = not assigned 8 = not assigned 9 = line pool A access code
Network Class of Service	Facility Restriction Level (FRL)	scheduled call routing
Network Bandwidth Management	Zone ID	Zone ID
		Virtual Private Network ID (VPNI)
	CS 1000 Zone ID = SRG50 Zone ID ZDP = VPNI. That is, the CS 1000 Zone ID is entered on the SRG50 in the Zone ID field, and the ZDP is entered on the SRG50 in the VPNI field (refer to "Bandwidth Management" on page 38).	
Trunks	public exchange	PSTN
IP telephone password	installer password	global password
	The two passwords can be made the same. Refer to "Registration password" on page 63.	

Normal and local mode overview: CS 1000 details

Normal mode

Telephones and clients that are connected to the SRG but are registered with the main office are in Normal Mode. The main office provides centralized call processing and extensive telephony features and applications for the SRG IP Phones. These telephones are registered to the main office TPS and are controlled by the Call Server at the main office. Survivable IP users connected through the SRG50 receive the features, 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

An IP Phone at the SRG may be in Local Mode for two different reasons;

1. IP Phone may have 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.
3. IP Phone is in Test Local Mode.

Devices that are physically located with the SRG and are controlled by the SRG system are said to be in Local Mode. These devices consist of analog telephones, analog devices such as, fax, and may include IP Phones. Normally IP Phones are registered to the main office, in Normal Mode, however when the IP Phone cannot reach the main office, it reverts to Local Mode.

IP Phone users in Normal Mode use the feature set on the main office. IP Phone users in Local Mode, receive only those features and tones that are provisioned on the SRG. Users of analog (500/2500-type) telephones always use the feature set on the SRG.

Survivability

SRG provides survivability against WAN failure, main office Call Server failure, main office Signaling Server failure, and Gatekeeper failure.

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 services based on a limited SRG feature set, which has significant differences from the CS 1000 software.

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's 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, it attempts to communicate with the SRG. Once it has established communications with the SRG, the SRG will redirect it to the main office. When the SRG IP Phone attempts to register with the main office, the SRG first queries the Primary NRS (NCS) for the main office Virtual Trunk node IP address to redirect the IP Phone. If the Primary NRS (NCS) is down or unreachable, the SRG queries the Alternate NRS (H.323 Gatekeeper), 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) 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 NRS fails and no Alternate NRS 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 the VoIP trunks are not available. (SRG analog [500/2500-type] telephones are accessible if alternate routing is available through the PSTN.)

Recovery to normal mode

If an IP Phone is in Local Mode due to WAN failure, or main office component failure, the SRG tries to communicate with the main office TPS at regular intervals. Once communication is established with the main office call server, the idle SRG IP Phones are automatically redirected and reregistered to the main office. IP Phones that were busy at the time communication was reestablished complete the call in Local Mode, and then reregister with the main office once 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.

> **IMPORTANT!**
>
> When a telephone or trunk in the main office calls a branch office IP telephone 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 voicemail or continuous ringback).

Virtual trunk capacity

The SRG50's capacity to support a number of simultaneous calls depends on the specific codec type used.

In normal mode, the codec selection used is controlled by specific programming of the CS1000. In this case: SRG50 supports up to a maximum of 15 Virtual trunks unless both the intrazone and interzone codecs are configured as Best Quality (G.711) in which case, the maximum number of virtual trunks would be 24.

In local mode, if the WAN has failed, there are no longer any virtual trunks available between the SRG50 and CS1000. However, the SRG50 will continue to convert calls from IP terminals for communication via the PSTN. In this case, if G.711 is used (recommended), the number of simultaneous calls from IP terminals to the PSTN supportable is a maximum of 24.

Vacant Number Routing

The SRG50 does not support Vacant Number Routing (VNR). Instead, the SRG50 uses Call Forward All Calls to emulate VNR for the IP telephones that are in normal mode. Call Forward All Calls is automatically cancelled when the phones revert to local mode.

A single destination code and route (or a group of destination codes and routes) can be configured on the SRG50 to route all calls not terminated locally by the SRG50. These calls are routed over the VoIP trunks. If the VoIP trunks become unavailable, the calls are routed to the proper location using PSTN fallback. This is similar to the VNR feature in Succession Branch Office.

Seamless dialing requires that the start digit of the DNs are unique for each system (coordinated dialing plan). If the start digit is the same on both systems, the local users on the SRG50 must dial a separate destination code before the main office DN.

Refer to "Setting up the private VoIP network" on page 75 for details on dialing plan and routing configuration.

Bandwidth Management

Three levels of bandwidth management are supported by the CS 1000:

- "Network Bandwidth Management (NBWM)"
- "Adaptive Network Bandwidth Management (ADBWM)"
- "Alternative Call Routing (ACR)"

Network Bandwidth Management (NBWM)

The SRG50 interoperates with the Network Bandwidth Management (NBWM) feature in a manner similar to MG 1000B, though only G.711 and G.729 codecs are supported. At the SRG50, a Virtual Private Network ID (VPNI) and Zone ID are entered with values defined by the main office configuration (refer to "Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR" on page 39). The VPNI and Zone ID allow the CS 1000 to recognize that H.323 calls to and from the SRG50 are from a specific Bandwidth Management zone.

NBWM allows bandwidth zones to be configured on a network basis so that codec selection and bandwidth allocation software can identify whether IP telephones or gateways are physically collocated (in the same bandwidth zone) even though they are controlled by different call servers. NBWM is used to define the codec selection policy and track bandwidth used for calls that traverse the WAN (interzone calls) and the LAN (intrazone calls). The bulk of configuration for NBWM is done at the main office.

Adaptive Network Bandwidth Management (ADBWM)

As with NBWM, only the VPNI and Zone ID are required at the SRG50 in order to implement he Adaptive Network Bandwidth Management (ADBWM) feature on the SRG50 (refer to "Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR" on page 39).

ADBWM uses real-time interaction to enhance the performance of Voice over Internet Protocol (VoIP) networks. ADBWM adjusts bandwidth limits and takes corrective action in response to Quality of Service (QoS) feedback. This adjustment occurs dynamically, while calls are in progress. A call server with ADBWM uses VPNIs and Zone IDs to keep track of the bandwidth being used between its own zone and zones belonging to other call servers. If the interzone QoS degrades below an acceptable level, the available bandwidth is reduced automatically between the two zones. When the QoS between the two zones improves, the bandwidth limit is allowed to return to normal.

Alternative Call Routing (ACR)

Configuration for Alternative Call Routing (ACR) at the SRG50 includes datafilling the Virtual Private Network ID (VPNI) and Zone ID required by NBWM and ADBWM. However, additional configuration is required and depends on the type of trunking provisioned at the main office: Attendant service or DID trunks (refer to "Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR" on page 39).

ACR for NBWM allows a station-to-station call (that is, a call that does not use a trunk) to overflow to traditional routes. Overflow can occur if there is insufficient interzone bandwidth available to carry the call, or if the QoS has degraded to unacceptable levels. The feature applies to station-to-station calls between a branch office and main office as well as from one branch office to another branch office, provided both stations are registered to the same main office.

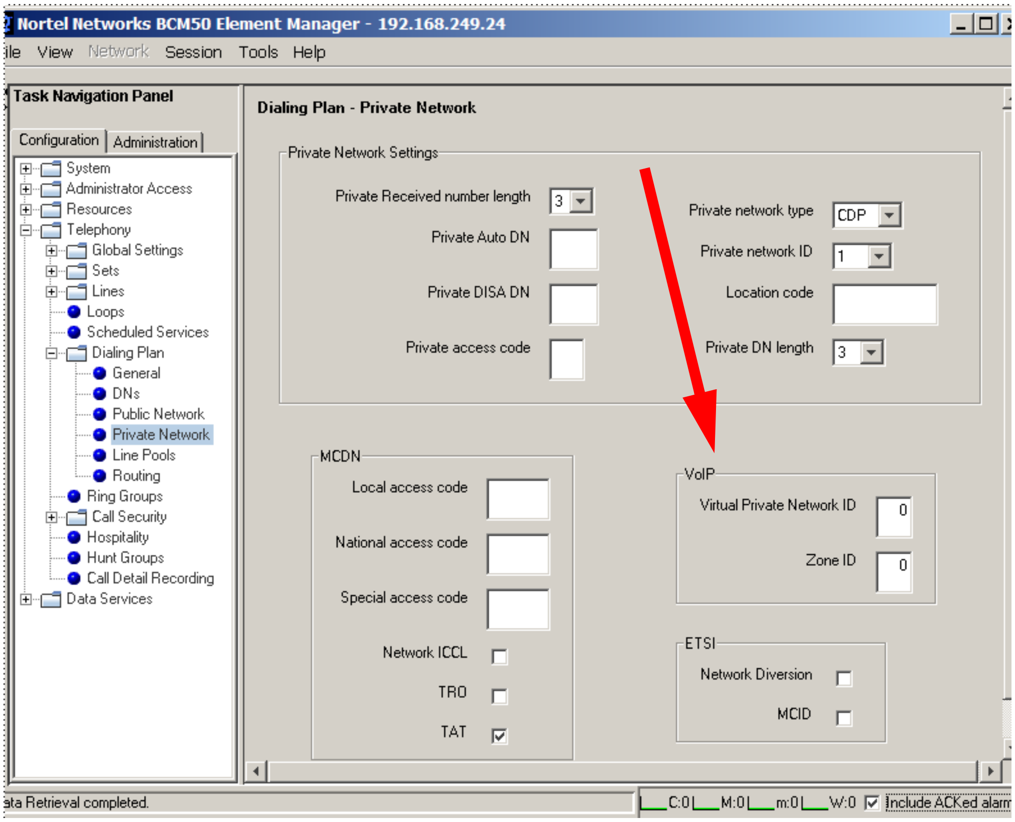
Network administrators who do not want calls to be blocked, yet have a limited amount of bandwidth available, can use ACR to overflow calls to conventional trunks (PSTN or Tie/MCDN). ACR allows calls to be routed by overflowing them, trading off the capital cost of WAN bandwidth against the incremental cost of overflowed calls.

Configuring the SRG50 for Bandwidth Management: NBWM, ADBWM, and ACR

To configure the SRG50 for NBWM and ADBWM

1. Obtain the Virtual Private Network ID and the Zone ID numbers configured at the main office.
2. Use the SRG50 Element Manager to enter these numbers in the appropriate fields at **Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network** panel **> VoIP** subpanel (Figure 4).

Figure 4 Dialing Plan - Private Network panel, VoIP subpanel



To configure Alternative Call Routing on the SRG50; with Attendant service

1 Complete the procedure: To configure the SRG50 for NBWM and ADBWM ("To configure the SRG50 for NBWM and ADBWM" on page 39).

2 Refer to "Call routing: configuring for outgoing calls" on page 86 and the *BCM50 Networking Configuration Guide* (N0027156) for more information on the logic behind this procedure.

3 Obtain the ALTPrefix for the SRG50 (configured at the main office).

4 Define a route to the main office Attendant over the PSTN.

- **a** Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab.
- **b** Add a new route (for example, 997).
- **c** Ensure that the **DN Type** is **Public (Unknown)**.
- **d** In the **External Number** field, enter the PSTN number of the main office Attendant telephone.
- **e** Assign the PSTN line pool to the route (select the line pool from the **Use Pool** pull-down menu; default is **A**).

5 Add a destination code.

a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab.

b Add a new destination code.

Use the ALTPrefix as the destination code.

c On the **ALTPrefix Destination Code** row, select the **Normal Route** field.

d Enter the route for the Attendant telephone (997).

e In the adjacent **Absorbed Length** field, select **All** from the pull-down menu.

When the SRG receives the ALTPrefix+DN digits from the main office, it looks up the destination code table, finds a match for the ALTPrefix, dumps all the digits (ALTPrefix+DN), and dials the main office Attendant.

To configure Alternative Call Routing on the SRG50; with DID trunks

Note: In order to support DID trunks, the MODN dialed by an SRG50 caller must match the DID digits.

Note: In the following procedure, it is assumed that the MODN matches the XXXX portion of the DID's NPA-NXX-XXXX; and that the ALTPrefix is 3 digits.

1 Complete the procedure: To configure the SRG50 for NBWM and ADBWM ("To configure the SRG50 for NBWM and ADBWM" on page 39).

2 Refer to "Call routing: configuring for outgoing calls" on page 86 and the *BCM50 Networking Configuration Guide* (N0027156) for more information on the logic behind this procedure.

3 Obtain the ALTPrefix for the SRG50 (configured at the main office).

4 Define a route for the NPA-NXXX portion of the main office DID numbers.

a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab.

b Add a new route (for example, 996).

c Ensure that the **DN Type** is **Public (Unknown)**.

d In the **External Number** field, enter the NPA-NXXX of the DID trunks that serve the main office.

e Assign the PSTN line pool to the route (select the line pool from the **Use Pool** pull-down menu; default is **A**).

5 Add a destination code.

a Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab.

b Add a new destination code.

Use the ALTPrefix as the destination code.

c On the **ALTPrefix Destination Code** row, select the **Normal Route** field.

d Enter the route for the route added above (996).

e In the adjacent **Absorbed Length** field, select **3** from the pull-down menu.

When the SRG receives the ALTPrefix+DN digits from the main office, it looks up the destination code table, finds a match for the ALTPrefix, dumps the 3- digit ALTPrefix, appends the DN to the **External Number** and dials the **External Number**+DN.

Configuring for Emergency Services Access (ESA)

The *Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50* guide covers the logic and procedures for configuring Emergency Services Access on both the SRG50 and the CS 1000. The information here fleshes out the SRG50 procedure in that guide.

Note: This procedure applies only to redirected IP phones when the SRG50 is in normal mode.

For IP telephones in local mode, and for other telephones at the SRG50, refer to the *BCM50 Networking Configuration Guide* (N0027156) for configuring emergency services.

To configure the SRG50 for Emergency Services Access

1 Verify that a remote access package has been assigned to the VoIP trunks (refer to"Remote Access Package for VoIP trunks" on page 88).

2 Obtain the ESA Special Number (SPN).

3 On the SRG50 Element Manager, access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and click the **Destination Codes** tab.

4 Add a destination code corresponding to the ESA SPN for the SRG50 branch office.

5 In the **Destination Codes** table, click the **Absorbed Length** field of the ESA SPN destination code. The numbers indicate the number of digits the SRG50 absorbs, from left to right.

6 Select the number of digits to absorb so that just the Emergency Services DN (ESDN) remains.

7 In the **Destination Codes** table, click the **Normal Route** field of the ESA SPN destination code. Enter a public route to the PSTN trunks.

> **Note:** The **Normal Route** field defaults to 000. Route 000 (click the **Routes** tab) is preconfigured to Use Pool A and cannot be changed. Pool A is preconfigured for PSTN trunks in the default state. Hence, if the default state of Pool A has not been changed, leave the Normal Route field as 000.
>
> To check the state of Pool A, navigate to **Configuration > Lines > Active Physical Lines**. Pool A must be assigned to at least one Trunk Type that provides access to the PSTN.
>
> Do not configure **Alternate Routes**.

8 Navigate to the Dialing Plan - Public Network panel (**Configuration > Dialing Plan > Public Network**).

9 In the **Public Network DN Lengths** subpanel, verify that there is a **DN Prefix** of 911 with a **DN Length** of 3.
If not, add the 911 **DN Prefix**. If required, double click the **DN Length** field to change it to 3.

CS 1000 information for the SRG50

In order to redirect IP telephones and forward calls to the main office (Call Forward All Calls feature), the SRG50 requires information about the main office network environment. This information is recorded through the SRG50 Element Manager on the **S1000 Main Office Settings** panel.

S1000 Main Office Settings panel

The following table lists and describes each field of the **S1000 Main Office Settings** panel. Record the actual value in the **Values** column to facilitate configuration and provide a record of the datafill.

Table 3 CS 1000 settings (Sheet 1 of 3)

Field	Values	Description
Primary Network Connect Server Address	<ip address>	IP address of the primary NCS.
Alternate Network Connector Server Address	<ip address>	IP address of the alternate NCS, if deployed. If not, enter the same address as for Primary Network Connect Server Address.

Table 3 CS 1000 settings (Sheet 2 of 3)

Field	Values	Description
Network Connect Server Port	16500 (default) Range: 0 to 65535	Port on the SRG50 used to connect to the NCS.
Heartbeat Protocol Port	16501 (default) Range: 0 to 65535	Port on the SRG50 that the SRG50 uses to monitor the status of the connection with the main office terminal proxy server (that is, to confirm connectivity with the main office)
VOIP Trunk Access Code		Access code for the main office VoIP trunk. Required for UDP dialing plan only. Ignored for CDP dialing plan, field can be left blank. VOIP Trunk Access Code = Destination code for VoIP trunks* = AC1** * Destination code for VoIP trunks is entered during configuration for advanced routing. Refer to "Call routing: configuring for outgoing calls" on page 86. ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>
Test Local Mode Timer	10 minutes (default) Range: 2 to 10 minutes	Period that an IP telephone remains in local mode after being set in local mode manually. Telephone returns to normal mode automatically at the end of the time-out. Local mode can be invoked by the Test Local Mode button on the telephone or by command from the main office.
H323 ID	SRG* (default) *This setting must be changed. Refer to the *BCM50 Networking Configuration Guide* (N0027156) for naming conventions.	Gatekeeper setting that identifies the SRG50. This value must match the value in the **Alias names** field of the Local IP gateway: **Configuration > Resources > Telephony Resources > Modules** panel **> Module type** column: select **IP Trunks > Details for Module: Internal details** panel **> Local Gateway** tab **> Gatekeeper Support** subpanel **> Alias names** field (refer to "Configuring VoIP trunking" on page 83).
Numbering Plan ID	Unknown ISDN/Telephony (E.164) Private Telephony (E.163) Telex (F.69) Data (X.121) National Standard Default: Private	The type of numbering plan at the main office.

Table 3 CS 1000 settings (Sheet 3 of 3)

Field	Values	Description
Type of Number	Unknown International Number National Number Special Number Subscriber Number ESN LOC (UDP) ESN CDP ESN Special Number Default: ESN CDP (for CDP dialing plans) (BUID = DN) UDP dialing plans: select ESN LOC (UDP) (BUID = LOC+DN)	The main office dialing plan. Ensure that the SRG50 private dialing plan is configured to match the selected value.
Node ID	9999 (default) Range: 0 to 9999	Automatically written to the IP telephone firmware when the IP telephone registers with the main office. Used to identify the node on the main office associated with the IP telephone DN.
MO Access Code Length	For CDP dialing plans: set to 0 For UDP dialing plans: set to length of line pool access code or destination code in front of LOC. Range: 0 to 34	The number of digits to add to the BUID (DN) so the main office system can determine if the incoming call is valid.

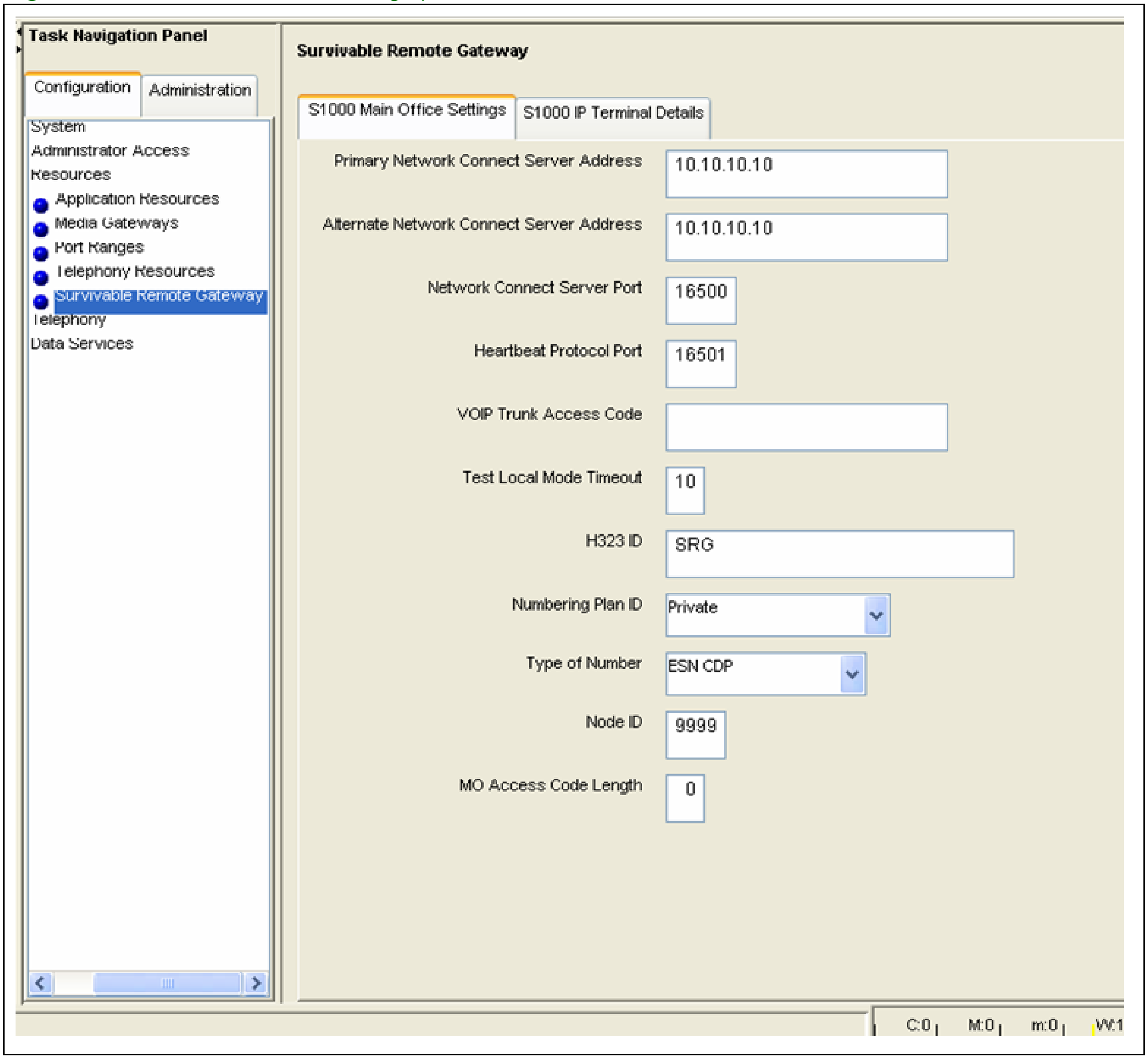
To datafill the S1000 Main Office Settings panel

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway** (Figure 5).

2 Select the **S1000 Main Office Settings** tab.

3 Enter the information in the appropriate fields.

Figure 5 S1000 Main Office Settings panel



Configuring IP telephones for redirection

Once an IP telephone at the SRG50 is configured (refer to "Setting up IP telephones" on page 63), it automatically registers with the SRG50 (S1). To configure an IP telephone for redirection to the main office call server, SRG-specific datafill is required. The SRG-specific configuration includes:

- "Recording numbers and models" on page 47
- "Datafilling the S1000 IP Terminal Details panel" on page 48
- "Redirecting the telephone to the main office call server" on page 49

Recording numbers and models

SRG50 DNs are assigned to IP telephones using the SRG50 Element Manager. Redirection to the main office requires configuration at the SRG50 to associate the SRG50 DN with the CS 1000 terminal number (TN) and the corresponding branch user ID (BUID).

When the CS1000 TN is configured, the IP telephone model forms part of the record. At the SRG50, the actual IP telephone configured to an SRG50 DN must be the same model that is configured in the TN record that is associated with the SRG50 DN.

The following table provides a record of the CS 1000 TN and BUID, the SRG50 MOTN and DN, and the model of phone, to facilitate installation and provide a permanent record of the configuration.

Table 4 IP telephone numbers and models (Sheet 1 of 2)

TN (CS 1000) MOTN (SRG50) (same number)	BUID (same number at CS 1000 and SRG50)	SRG50 DN	IP telephone model

Table 4 IP telephone numbers and models (Sheet 2 of 2)

TN (CS 1000) MOTN (SRG50) (same number)	BUID (same number at CS 1000 and SRG50)	SRG50 DN	IP telephone model

Datafilling the S1000 IP Terminal Details panel

The SRG50 Element Manager provides SRG-specific panels for recording the CS 1000 TN and BUID that are associated with a particular SRG50 DN. The following table lists and describes the fields on the **S1000 IP Terminal Details** panel.

Table 5 SRG_S1000Terminal Details fields

Field	Values	Description
DN	Read-only	The SRG50 DN assigned to the telephone. The SRG50 DN must be configured before proceeding with the procedures that follow in this section. Refer to "Configuring telephone (DN) records" on page 66.
H/W ID	Read-only	Hardware ID. Unique for each IP telephone.
Status	Read-only	Current status of the telephone. Refer to "IP terminal details" on page 96 (expand the field to read the entire status message).
F/W Version	Read-only	Updated by the main office when a terminal is sent back to the SRG50 for firmware upgrade purposes. The field specifies the firmware version required by the main office.
MOTN	XXX	Required for telephone redirection. The field is the main office TN associated with the IP telephone.
BUID	CDP network: <DN> UDP network: <VoIP access code> + <LOC> + <DN>	Required for telephone redirection. The field represents the dialable number of an IP telephone at the SRG50 when it is called from a phone located at the main office or another branch office. The BUID at the SRG50 must be the same as the BUID at the main office.
MO TPS	Read-only	This field echoes the address of the main office terminal proxy server when the IP telephone is redirected.

Note: The SRG50 DNs must be configured before the following procedures can be undertaken. Refer to “Configuring telephone (DN) records” on page 66.

To enter the MOTN and BUID

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.

2 Select the **S1000 IP Terminal Details** tab (Figure 6).

3 Refer to the table of IP telephone numbers and models recorded in the table provided earlier in this section.

4 Select the required DN.

5 Press the **Modify** button.

6 Enter the MOTN and the BUID in the appropriate fields.

Redirecting the telephone to the main office call server

To redirect the telephone to the main office call server

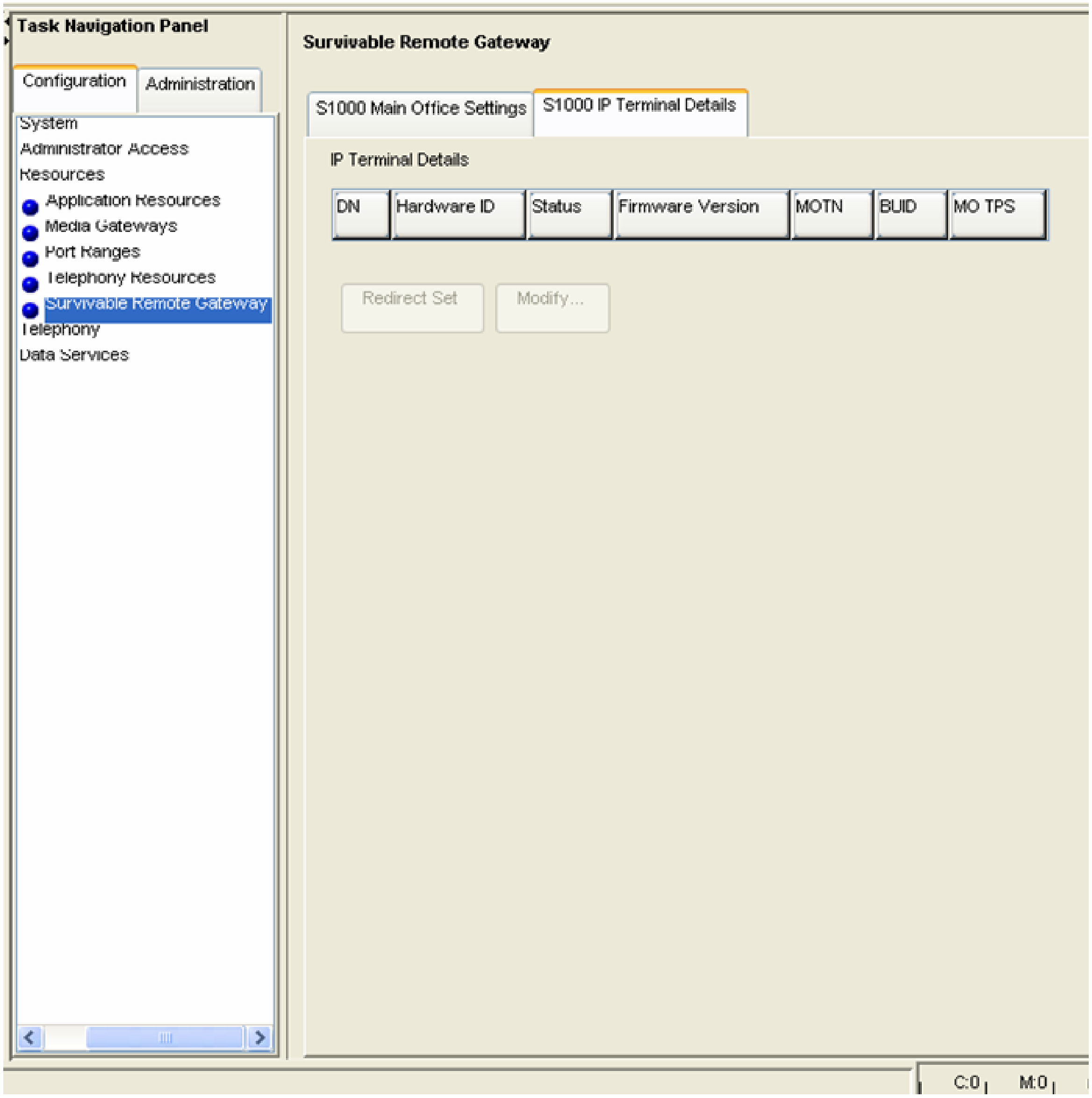
1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.

2 Select the **S1000 IP Terminal Details** tab (Figure 6).

3 Select the DN of the telephone to be redirected.

4 Press the **Redirect Set** button.

Figure 6 S1000 IP Terminal Details panel



IP telephone settings

For IP telephones that are redirected to the main office call server, incorporate the settings shown in the following table. The *BCM50 IP Telephone Installation and Configuration Guide* (N0027269) and the *BCM50 Networking Configuration Guide* (N0027156) provide detailed instructions for configuring IP telephones.

Table 6 Configuration settings for redirected IP telephones

Parameter	Setting
S1 IP	SRG50 IP address
S1 Port	7300
S1 Action	1
S1 Retry Count	1
S2 IP	SRG50 IP address
S2 Port	7300
S2 Action	1
S2 Retry Count	1

Firmware upgrade

The redirected IP telephones at the SRG50 are under the control of the main office call server for the majority of their deployment and receive all of their features in that context. Therefore, the version of IP set firmware must align with the requirements of the CS 1000.

Firmware supported by SRG50 Release 1.0

The following table lists the IP clients and related firmware versions supported on the SRG50. The SRG50 column indicates the firmware versions included with the SRG50 software. The CS 1000 columns identify the version of firmware to use for specific releases.

IP Client	SRG50	CS 1000	
		Release 3.0 **Release 4.0**	**Release 4.5** (apply via Enhanced Firmware Download feature)
Phase I: 2002, 2004	B65 or greater	B65 or greater	B76 or greater
Phase II: 2001, 2002, 2004	D98 or greater	D98 or greater	D98 or greater
2007	C23 or greater	C23 or greater	C23 or greater
2210/2211 WLAN Handsets	Not embedded in SRG50 software.	Use the 2210/2211 GA firmware load.	

Release 4.5 firmware upgrade procedure

When an IP telephone requires a firmware upgrade, the CS 1000 uses the umsUpgradeAll command, or variant, to redirect the telephone back to the SRG50 for upgrading. If the required file does not exist on the SRG50, or its version is incorrect, the SRG initiates an FTP session to the TPS for that phone to retrieve the required file. The SRG50 upgrades the phone and redirects it back to the CS 1000.

Chapter 4
CS 2000 considerations

The following activities are specific to the interoperability of the Nortel Centrex IP Client Manager (CICM) gateway in a Communication Server 2000 (CS 2000) main office with an SRG50:

- "Hardware and software requirements" on page 53
- "Supported IP telephones" on page 53
- "Restrictions and limitations" on page 54
- "Configuration considerations for CICM interoperability with SRG50" on page 54
- "Configuring IP telephones for redirection" on page 55
- "Registering the IP telephones at the SRG50: Test Local Mode" on page 61
- "Firmware upgrade" on page 61

An overview of CICM interoperability with the SRG50 is provided in the solution-level Basics document associated with your CS 2000-based solution.

Note: Currently, the SRG50 is a First Customer Application for the CS 2000 and is working through the Nortel Verification Office to achieve full general availability in conjunction with the Centrex IP Client Manager solution.

Hardware and software requirements

For CICM interoperability with the SRG50, the following hardware and software are required:

- a fully configured 5370 or 5385 CICM gateway with Active Call Failover, and a 5370 or 5385 CICM Manager in the CS 2000 main office running the SN08 release
- a fully configured BCM50 in the SRG branch office with the SRG50 software update and keycode applied

Supported IP telephones

The following IP telephones are supported for CICM interoperability with the SRG50:

- Phase I IP Phones 2002 and 2004 with firmware 1.74 and later
- Phase II IP Phones 2001, 2002, and 2004 with firmware 3.92 and later

Restrictions and limitations

The following restrictions and limitations apply to CICM interoperability with the SRG50:

- SRG50 is not supported as an H.323 gateway to the CS2000 main office.
- The m6350 softclient and IP Phone 2033 are not supported.
- Security is not available in local mode, which is when the IP telephone is connected to the SRG50.
- Registering IP telephones as redirected sets at the SRG50 must be done manually.

Configuration considerations for CICM interoperability with SRG50

The IP telephones located at the SRG50 branch office must have the following configuration settings:

Field	Value
S1 IP	<CICM IP address>
S1 Port	5000
S1 Action	1 (non-secure) or 6 (secure)
S2 IP	<SRG50 IP address>
S2 Port	7300
S2 Action	1 (non-secure)

Refer to the *Centrex IP Client Manager Etherset Installation Guide and User Manual, NN10027-113*, for detailed instructions on how to install and configure IP telephones.

The IP telephones that are located at the SRG50 branch office, initially register to the CICM gateway (S1) located in the main office.

Note: If the CICM gateway to which the IP telephone is connected is part of an enterprise profile, and the user selects to connect to a different CICM gateway, the set is redirected to the remote CICM gateway (S3). If the remote CICM gateway does not respond, the set falls back to the default CICM gateway (S1). If the remote CICM gateway does respond and the change is committed, it replaces the default CICM gateway (S1).

Configuring IP telephones for redirection

Once an IP telephone at the SRG50 is configured (refer to "Setting up IP telephones" on page 63), it automatically registers with the CICM. In the case where the VoIP trunks to the main office become unavailable, IP telephones at the SRG50 revert to local mode IF the telephones are registered with the SRG50 and are configured for redirection.

This section describes the activities required to configure the telephones for redirection. The next section, "Registering the IP telephones at the SRG50: Test Local Mode" on page 61, explains how to register the telephones with the SRG50.

The two configuration paths for redirection are:

- Call forwarding of calls coming into the SRG50 from the local PSTN is required (that is, the SRG50 is equipped with PSTN trunks and you want calls from the PSTN to be call forwarded automatically to the main office DN provisioned for the IP telephone at the SRG50).

Note: Call forwarding is not supported in SN08.

- Call forwarding is not required.

Step 1: Enter the CS 2000 main office settings

The following table lists and describes the two fields of the **CS2000 Main Office Settings** panel. Record the actual value in the **Values** column to facilitate configuration and provide a record of the datafill.

Table 7 CS 2000 settings

Field	Value	Description
VOIP Trunk Access Code		Access code for the main office VoIP trunk. Only required when call forwarding is required. Only required for UDP dialing plan. (Ignored for CDP dialing plan, recommend that field is left blank.) VOIP Trunk Access Code = Destination code for VoIP trunks* = AC1** * Destination code for VoIP trunks is entered during configuration for advanced routing. Refer to "Call routing: configuring for outgoing calls" on page 86. ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>
Test Local Mode Timer	10 minutes (default) Range: 2 to 10 minutes	Period that an IP telephone remains in local mode after being set in local mode manually, or if redirected to the SRG50 because connectivity to the main office is lost. Telephone returns to normal mode automatically at the end of the time-out (given that the VoIP trunks are available). Use the Test Local Mode button on the telephone to invoke local mode manually. Test Local Mode can be terminated by pressing the Exit button at the telephone or using the Redirect Set button on the SRG50 Element Manager (**Configuration > Resources > Survivable Remote Gateway > CS2000 IP Terminal Details** tab).

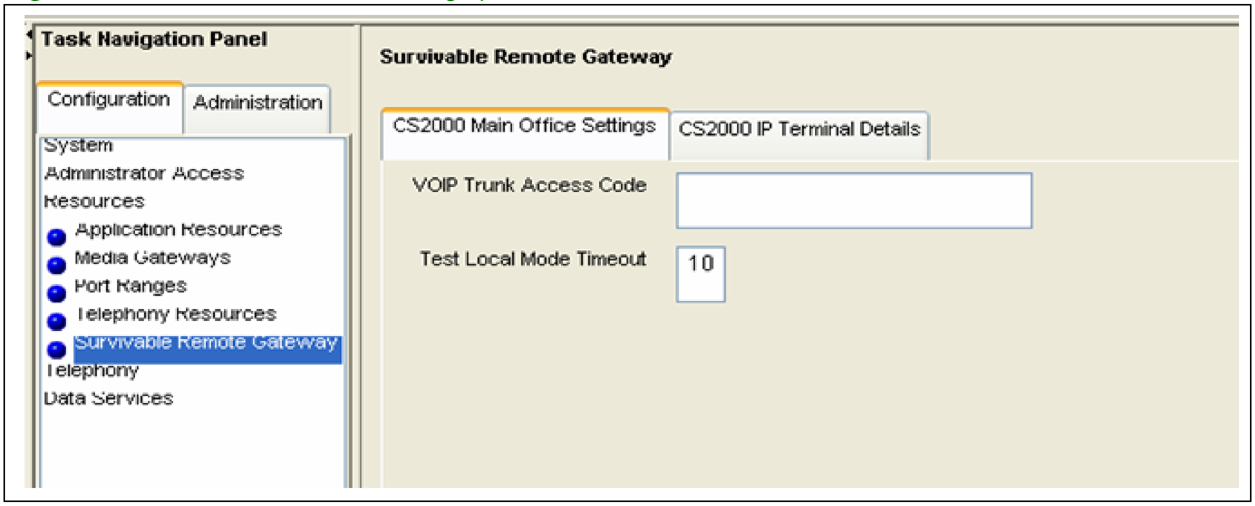
To datafill the CS2000 Main Office Settings panel

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.

2 Select the **CS2000 Main Office Settings** tab (Figure 7).

3 Enter the information in the appropriate fields.

Figure 7 CS2000 Main Office Settings panel



Step 2: Record numbers and models (call forwarding only)

Telephone models are hard-coded to the telephone numbers at the main office. In normal mode, IP telephones* at the SRG50 are registered with the main office, and the main office provides all call handling. In order for calls coming into the SRG50 from the local PSTN to be call forwarded to the main office, the model of each IP telephone installed at the SRG50 must match the model of phone configured for the corresponding number at the main office. This requires mapping the CS 2000 main office directory number (MODN) to the SRG50 DN. When the following table is completed, it provides a record of the model of the IP telephone, the SRG50 DN, and the CS 2000 MODN.

* Provided that the IP telephone has been configured to operate in SRG mode. Refer to "Configuration considerations for CICM interoperability with SRG50" on page 54.

Note: The MODN is the DN of the telephone when it is registered at the CICM gateway.

Table 8 Mapping IP telephones to MODNs and SRG50 DNs (Sheet 1 of 2)

MODN	SRG50 DN	IP telephone model

Table 8 Mapping IP telephones to MODNs and SRG50 DNs (Sheet 2 of 2)

MODN	SRG50 DN	IP telephone model

Step 3: Configure the MODN (call forwarding only)

CS2000 IP Terminal Details panel

The MODN is configured on the **CS2000 IP Terminal Details** panel. The following table lists and describes the fields on this panel.

Table 9 SRG_CS2000Terminal Details fields (Sheet 1 of 2)

Field	Values	Description
DN	Read-only	The SRG50 DN assigned to the telephone. This must be configured before proceeding with the procedures that follow in this section. Refer to "Configuring telephone (DN) records" on page 66.
H/W ID	Read-only.	Hardware ID. Unique for each IP telephone.

Table 9 SRG_CS2000Terminal Details fields (Sheet 2 of 2)

Field	Values	Description
Status	Read-only	Current status of the telephone. Refer to "IP terminal details" on page 96 (expand the field to read the entire status message).
Main Office DN (MODN)		Required for call forwarding. Leave blank if call forwarding is not required. If provisioned, the SRG50 invokes a Call Forward All Calls to the MODN when the IP telephone is redirected to the main office. Call forwarding not supported in SN08, leave field blank.
MO TPS	Read-only	This field echoes the IP address of the CICM to which the set is connected. That is, the IP address in S1.

Note: The SRG50 DNs must be configured before the following procedures can be undertaken. Refer to "Configuring telephone (DN) records" on page 66.

To enter the MODN

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.

2 Select the **CS2000 IP Terminal Details** tab (Figure 8).

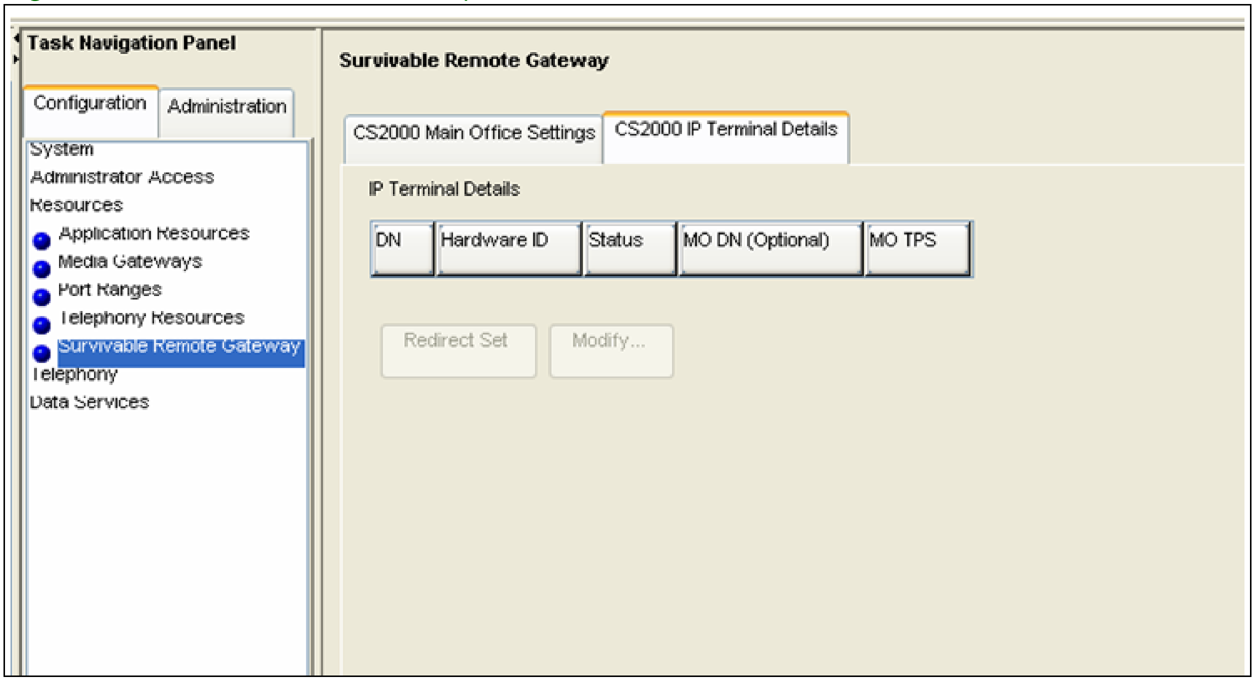
3 Refer to the information recorded in "Step 2: Record numbers and models (call forwarding only)" on page 57.

4 Select the required DN.

5 Press the **Modify** button.

6 Enter the MODN in the **MODN** field.

Figure 8 CS2000 IP Terminal Details panel



Step 4: Redirect the telephone to the CICM

To redirect the telephone to the CICM

1. On the SRG50 Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.
2. Select the **CS2000 IP Terminal Details** tab (Figure 8).
3. Select the DN of the telephone to be redirected (this is the DN that appears on the telephone when in local mode).
4. Click the **Redirect Set** button.

Registering the IP telephones at the SRG50: Test Local Mode

When the IP telephones are configured (refer to "Setting up IP telephones" on page 63), they automatically register with the CICM (S1). To register the IP telephones at the SRG50 (S2), use the Test Local Mode option to redirect the phone to the SRG50. The Test Local Mode option is available under the Diagnostics menu of the supported IP telephones.

Note: The Diagnostics menu option is available on the supported IP telephones only if the Disable Diagnostics Menu parameter on the User Profile and Profile overrides pages of the CICM Element Manager is set to No. If the Disable Diagnostics Menu parameter is set to Yes, the Diagnostics menu option is not present on the user's main menu. To view or to change the Disable Diagnostics Menu parameter, refer to the CICM Configuration Management document, NN10240-511.

The first time that the telephone is redirected to the SRG50, the telephone remains connected to the SRG50 until it is configured at the SRG50 as a redirected set and the **Redirect Set** button is pressed (refer to "Configuring IP telephones for redirection" on page 55 for details).

On subsequent occasions, you can exit Test Local Mode by waiting for the feature to time out or by pressing the key with the Exit button ( ). This button is active only when the telephone is in the local mode test. If the phone does not have an Exit button, you must wait until the test times out.

Note: If an IP telephone is redirected to the SRG50 because the connection to the main office has become unavailable, the SRG50 does not attempt to redirect the phone back to the main office until Test Local Mode times out. This is true even if connectivity to the main office is reestablished before the test times out. Redirection to the main office can be invoked manually by pressing the Exit button.

Firmware upgrade

The redirected IP telephones at the SRG50 are under the control of the main office call server for the majority of their deployment and receive all of their features in that context. Therefore, the version of IP set firmware is aligned with the requirements of the CS 2000.

Chapter 5
Setting up IP telephones

IP telephone setup and DN configuration are described in detail in the *IP Telephone Installation and Configuration Guide* and the *BCM50 Networking Configuration Guide*, respectively. SRG-specific procedures and settings include:

Registration password

If a registration password is configured on the SRG50, the IP telephone installer must enter the password before the telephone can be configured.

To set the IP telephone registration password

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

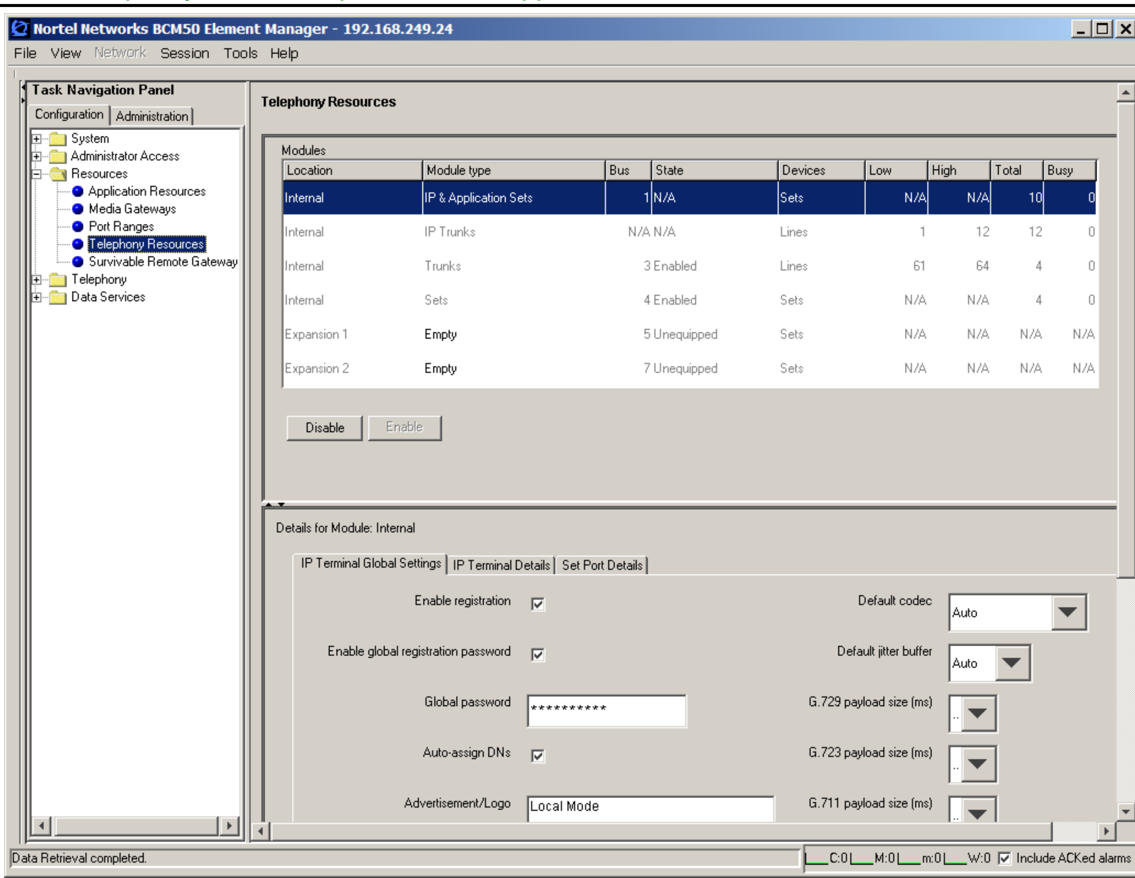
2 Select the **IP & Application Sets** row (Figure 9).

3 On the **Details for Module: Internal** subpanel, select the **IP Terminal Global Settings** tab.

There are three fields that define the password registration process: **Enable registration**, **Enable global registration password**, and **Global password**.

Enable registration	Must be selected to allow IP telephones to register.
Enable global registration password	Select if a password is going to be entered in the next field, **Global password**. If selected, the installer must enter the Global password (below) at the IP telephone before the telephone can be configured.
Global password	Enter the password. If no password is entered, or the **Enable global registration password** is not selected, no password is required to configure the IP telephone. Note: If the main office is a CS 1000, this password can be coordinated with the CS 1000 configuration.

Figure 9 Telephony Resources panel, IP and Application Sets



4 Select and enter the values to meet the password requirements of your installation.

5 Set the **Auto-assign DNs** check box according to the requirements of your installation (if set, the SRG50 automatically assigns DNs; refer to *BCM50 Device Configuration Guide* for details).

Local mode indication

When an IP telephone is in local mode, a message is displayed on the phone to indicate the local mode state to the user. The default setting is **Local Mode**.

To change local mode indication

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

2 Select the **IP & Application Sets** row (Figure 9).

3 On the **Details for Module: Internal** subpanel, select the **IP Terminal Global Settings** tab.

4 The **Advertisement/Logo** field specifies the message that provides local mode indication. Change as required.

IP telephone codec and jitter settings

When the IP telephones are operating in local mode, codec and jitter settings are set on a phone-by-phone basis. Configure the settings to meet the requirements of the local SRG50 environment. They do not have to be the same as the main office settings (in contrast to the QoS settings for the VoIP trunks; refer to "QoS settings (codec, jitter buffer, and related items)" on page 80).

To enter codec and jitter settings for IP telephones in local mode

1 On the SRG50 Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

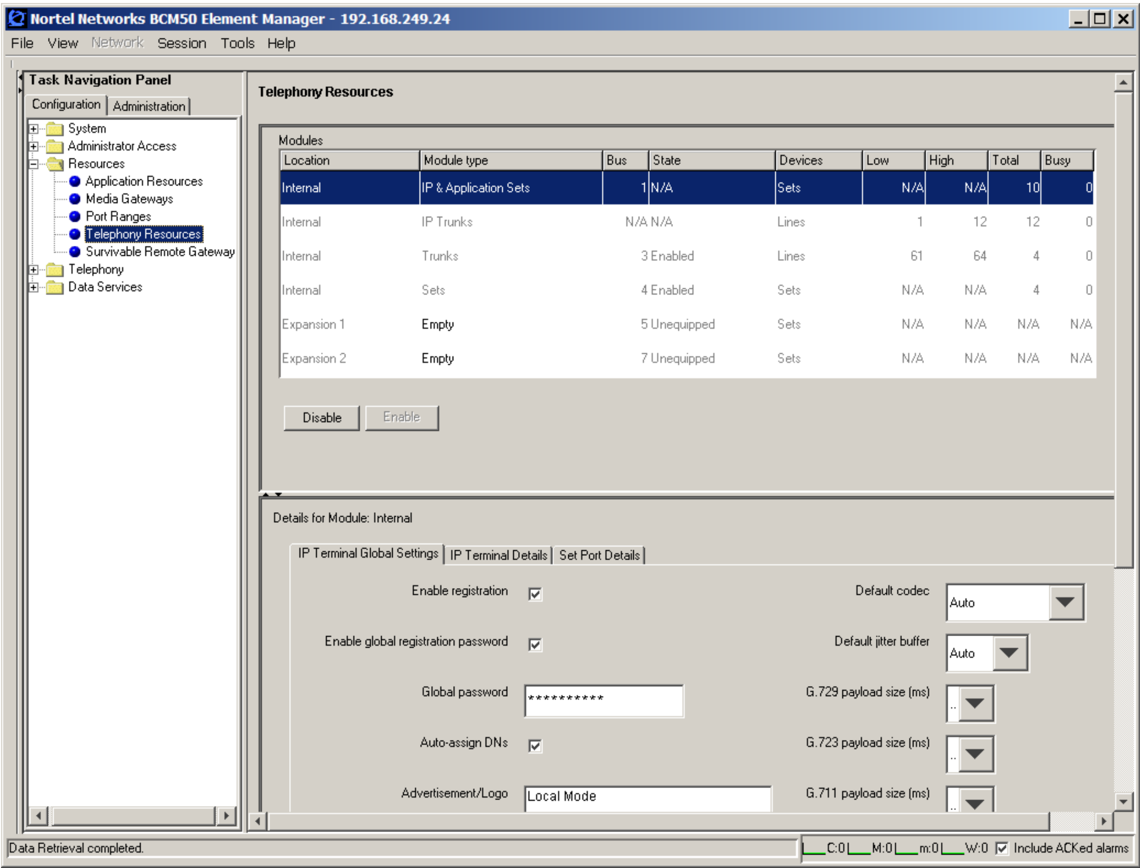
2 On the **Modules** panel, locate the **Module type** column and select the **IP & Application Sets** row (Figure 10).

3 On the **Details for Module: Internal** subpanel, select the **IP Terminal Global Settings** tab.

The fields related to QoS are on the right side of the panel.

4 Enter the appropriate values.

Figure 10 Modules panel, IP & Application Sets



Configuring telephone (DN) records

DN records for IP telephones are configured through the **All DNs** panel (**Configuration > Telephony > Sets > All DNs** (Figure 11). The *BCM50 Networking Configuration Guide* and the *BCM50 Device Configuration Guide* provide basic instructions for configuring DNs and IP telephones. The following instructions are in addition to these instructions and only apply to IP telephones that are to be redirected to a main office call server.

Note: It is assumed that the line pools have been assigned. In the default configuration, VoIP trunks are assigned to line pool BlocA and the four PSTN trunks are assigned to line pool A. Refer to "Creating the SRG50" on page 10, "SRG50 terminology" on page 15, and "Foundation configuration" on page 27 for more information.

To configure DN records for redirected IP telephones

1 On the SRG50 Element Manager, navigate to the **All DNs** panel (**Configuration > Telephony > Sets > All DNs**).

2 Select the **Line Access** tab.

3 Identify the row of the DN record to be configured.

4 Refer to the list of numbers and phone models recorded in the table in the server-specific chapter (“CS 1000 considerations” on page 33 and“CS 2000 considerations” on page 53).

5 From the pull-down menu, select the **Model** of telephone assigned to this DN.

> **Note:**
> For CS 1000: Refer to “Recording numbers and models” on page 47.
>
> For CS 2000: Refer to “Step 2: Record numbers and models (call forwarding only)” on page 57. (If the call forwarding feature is not implemented, select **i2004/i2050** for all DNs.)

6 To support outgoing caller ID over the VoIP trunk, the Private OLI field (**Priv. OLI**) must be set to the DN.

7 To support outgoing number display over the PSTN, enter the public access number for the telephone in the Public OLI field (**Pub. OLI**).

8 Leave the **Fwd All** field blank to disable the Call Forward All Calls feature when the telephone is in local mode. In normal mode, the SRG50 can forward all calls to the main office call server automatically (refer to “Call forwarding options” on page 71). In local mode, the Call Forward All Calls feature is automatically discontinued and the SRG50 routes calls to the SRG50 DNs.

Call Forward All Calls does not require **Allow redirect** to be enabled (**Allow redirect** is found on the **Capabilities** tab of the **Details for DN** subpanel when the **Capabilities and Preferences** tab is selected).

9 To assign specific PSTN lines to each telephone, add the line(s) (**Details for DN** details panel > **Line Assignment** tab). You would do this if, for example, you want one user to field all customer calls when the system is in local mode.

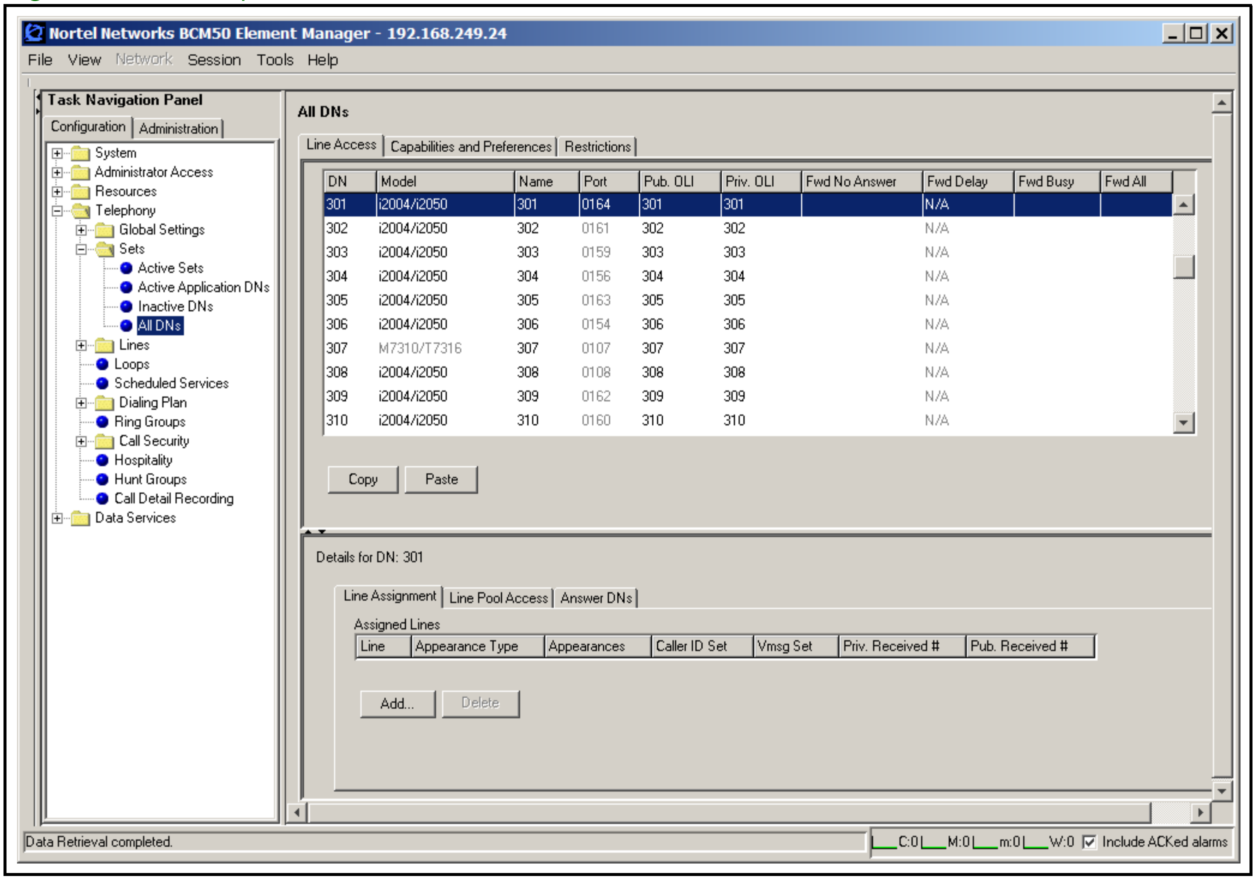
10 Ensure that the **Appearance Type** (**Details for DN** details panel > **Line Assignment** tab) is set to **Ring only**.

11 Assign the VoIP and PSTN trunk line pools to the DN (**Details for DN** details panel > **Line Pool Access** tab).

12 Assign a target line to the DN (**Details for DN** details panel > **Line Pool Access** tab). Refer to “SRG50 terminology” on page 15 for a description of target lines.

> **Note:** At this point, you may want to configure the received numbers. Refer to “Configuring received numbers” on page 69 and then return to this procedure.

Figure 11 All DNs panel



13 Select the **Capabilities and Preferences** tab.

14 Set **Intercom Keys** to 1 if required.

15 On the **Details for DN** subpanel, select the **Button Programing Table** tab.

16 Program the voice mail access button with the PSTN dialup for the main office voice message system.

Model	Button
2001	Message
2002	06
2004	08
2007	08
2050, 2050CE	08
2210	08
2211	08

Configuring received numbers

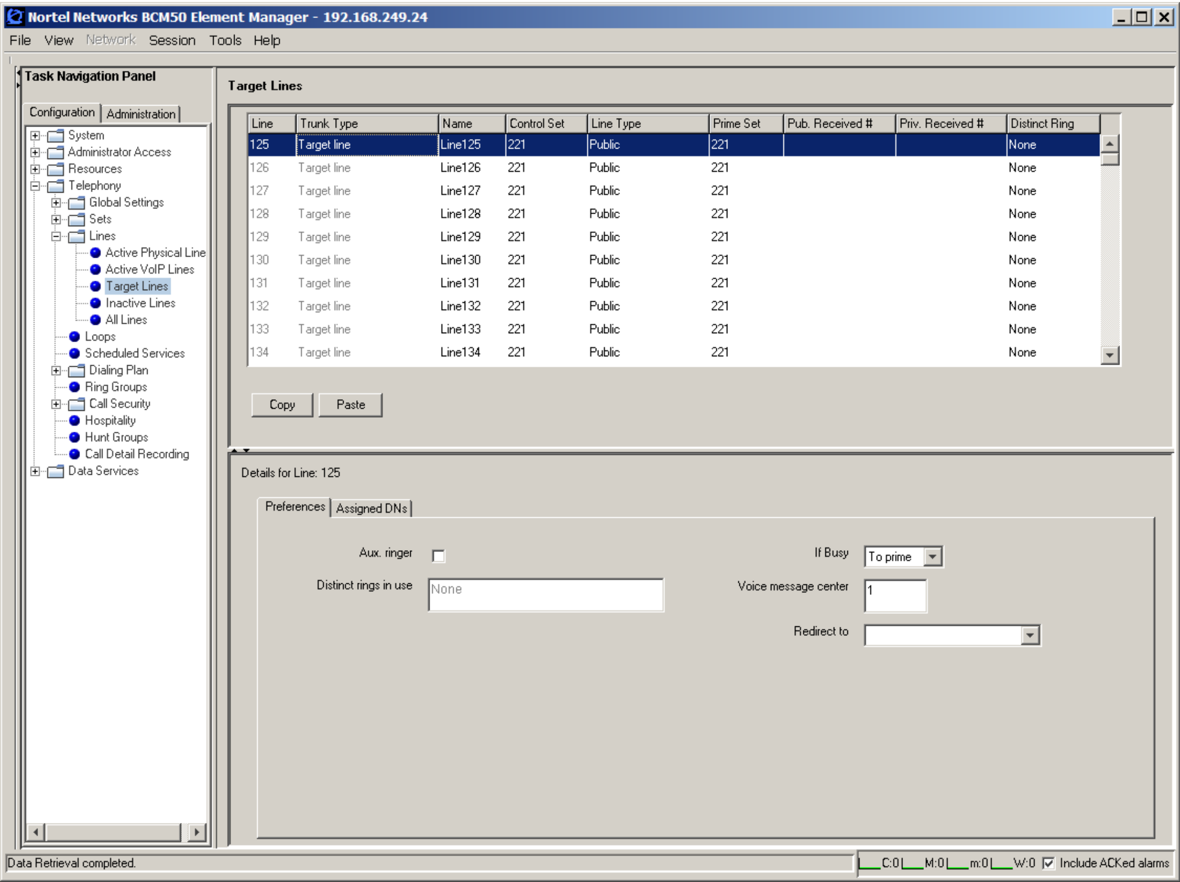
The **Public Received number length** and the **Private Received number length** (refer to "Basic parameters" on page 77) determine the number of digits that the SRG50 retains for call processing. The retained digits are mapped to the DN using fields provided on the **Target Lines** panel. (For more information on target lines, refer to "SRG50 terminology" on page 15).

Note: If the retained digits are the same as the DN, the fields (**Pub. Received #** field and **Priv. Received #** field) can be left blank.

To map received numbers to the DN

1. On the SRG50 Element Manager, navigate to **Configuration > Telephony > Lines > Target Lines** (Figure 12).
2. Select the target line of the DN for which you want to configure the received numbers.
3. In the **Pub. Received #** field, enter the retained received digits for calls originating from the PSTN.
4. In the **Priv. Received #** field, enter the retained received digits for calls originating from the private network (that is, the VoIP trunks). This number is usually the same as the DN.

Figure 12 Target Lines panel



Configuring DHCP settings

To configure DHCP settings for SRG operation

1. On the SRG50 Element Manager, navigate to **Configuration > Data Services > DHCP Server**.
2. Select the **General Settings** tab.
3. On the **DHCP Server is** pull-down menu, select **Enabled - IP Phones Only**.
4. On the **WINS node type** pull-down menu, select **H-node**.
5. In the **Default gateway** field, enter an address that meets the requirements of the SRG50 LAN.
6. In the **Lease time** field, enter a value that meets the requirements of your system.

   Leave all other fields under the General Settings field blank.
7. Select the **IP Terminal DHCP Options** tab.

8 On the **Primary Terminal Proxy Server (S1)** and **Secondary Terminal Proxy Server (S2)** subpanels:

a In the **IP address** field, enter the IP address of the SRG50.

b On the **Port** pull-down menu, select **SRG**.

c In the **Retry** field, enter the number of retries that the IP telephone is allowed to connect to the SRG50 before an event is generated (refer to "IP terminal details" on page 96).

9 Select the **Address Ranges** tab and add a range of IP address to meet the requirements of your system.

Call forwarding options

There are two options for configuring call forwarding on the SRG50:

- The target DN is determined by the BUID (CS 1000) or the MODN (CS 2000).

  In this case, the information required for call forwarding is entered using the SRG-specific panels of the SRG50 Element Manager (**Configuration > Resources > Survivable Remote Gateway**). Refer to the server-specific chapters for details ("CS 1000 considerations" on page 33 and"CS 2000 considerations" on page 53).

  Note: Call forwarding is mandatory for the CS 1000 but is optional for the CS 2000.

- The target DN is configured explicitly for each IP telephone.

  The DN is configured through the SRG50 Element Manager on the **Telephony > Sets > Active Sets** panel using the standard BCM50 procedure (refer to the *BCM50 Device Configuration Guide* for details). Typically, the call forward number is the BUID (CS 1000) or MODN (CS 2000).

  The disadvantage of the second option is that the installer must configure the target DN in two places: on the **Active Sets** panel and on the SRG50-specific panels (the SRG-specific panels **must** be completed). The SRG50 looks at the SRG50-specific panels first. It goes to the **Active Sets** panel only if the **VOIP Trunk Access Code** has not been configured.

Configuration settings for redirected phones

The *BCM50 IP Telephone Installation and Configuration Guide* (N0027269) and the *BCM50 Networking Configuration Guide* (N0027156) provide detailed instructions for configuring IP telephones. For IP telephones that are redirected to the main office call server, incorporate the settings shown in the following table.

Table 10 Configuration settings for redirected IP telephones

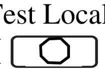
Parameter	CS 1000	CS 2000
S1 IP	SRG50 IP address	Default CICM Node Address
S1 Port	7300	Default CICM Node Port
S1 Action	1	1 (non-secure) or 6 (secure)
S1 Retry Count	1	1 (non-secure) or 6 (secure)
S2 IP	SRG50 IP address	SRG50 IP address
S2 Port	7300	7300
S2 Action	1	1
S2 Retry Count	1	1

Refer to the server-specific chapters ("CS 1000 considerations" on page 33 and"CS 2000 considerations" on page 53) for more details.

Test Local Mode

An IP telephone operating in normal mode can be forced to redirect to the SRG50. This allows the telephone user, and system administrator, to test local mode operation without taking down the VoIP trunk to the main office.

Invoking local mode test is specific to the main office call server. Refer to the server-specific chapters for details (for CS 1000: "S1000 Main Office Settings panel" on page 43; for CS 2000: "Registering the IP telephones at the SRG50: Test Local Mode" on page 61).

Generally, you exit Test Local Mode by waiting for the feature to time out or by pressing the key with the Exit button (). This button is active only when the telephone is in the local mode test. If the phone does not have an Exit button, you must wait until the test times out.

Note: For Nortel 2210 and 2211 WLAN handset, pressing the End key causes the phone to exit Test Local Mode.

Features in local mode

In local mode, IP telephones at the SRG50 no longer have access to the full suite of main office applications. However, the SRG50 does provide a set of features that include connectivity with the local PSTN, access to Emergency Services, and the ability to call local extensions.

The SRG also supports the following features in local mode:

- Hold
- Transfer (dedicated key on the 2002, 2004, and 2007 models)
- Call Forward No Answer/Busy (if the feature has been enabled on the DN: C**onfiguration > Telephony > Sets > All DNs**)
- Last Number Redial (dedicated key on the 2002/2004 models)
- Inbox Key (on 2002/2004 models)

The user experience in local mode can be enhanced if certain global feature settings are coordinated with the main office so that the settings are the same at both the main office and the SRG50. These feature settings are configured with the **Feature Settings** panel on the SRG50 Element Manager interface (**Configuration > Telephony > Global settings > Feature Settings**). Feature settings that can be coordinated with the main office are:

- **Background music** (if it is provided for on-hold)
- **On-hold**
  This determines if a caller on hold hears tones, music, or nothing.
- **Receiver volume**
  Set to use the system volume, since IP users cannot use the feature code to set a default telephone volume.
- **Delayed ring transfer**
  If a transfer to an external number is not answered, you can indicate if the call will be dropped (Off) or transferred to the designated Prime telephone.

  Check the **Transfer callback timeout**. This setting defaults to **After 4 rings**. If you are using the **Delayed ring transfer** feature, turn **Transfer callback timeout** off if you want all unanswered transferred calls to ring at the Prime set (usually the system attendant). If you want the transferred call to ring at the telephone from which it was transferred first, set this field to a setting that is less than the setting for **Delayed ring transfer**.
- **Held line reminder**
  If set to a time, determines period between when a call is put on hold and when a short tone sounds at the telephone to indicate the call is still on hold.
- **Alarm set**
  Enter a DN to a two-line analog telephone, since the IP telephones will not be able to access the alarms, or set to **None** if you do not want to use an alarm set on the system.
- **Language and Contrast**
  Language and Contrast are DN-specific settings and are configured at **Configuration > Telephony > Sets > All DNs > All DNs** panel **> Capabilities and Preferences** tab **> Details for DN** subpanel **> Preferences** tab.

Features not supported in local mode include: Hot Desking, Do Not Disturb, Page, Call Forward, Background Music, Call Park, Call Pickup, Speed Dial, and Conference.

911 Emergency Services Support

For IP telephones in local mode, and for other telephones at the SRG50, the *BCM50 Networking Configuration Guide* (N0027156) provides details for configuration of 911 emergency services.

For redirected IP telephones in normal mode, the IP telephone is registered with the main office call server. Ensure that the main office call server is configured so that a 911 call from an IP telephone at the SRG50 is routed back to the SRG50's local PSTN. "Configuring for Emergency Services Access (ESA)" on page 42 includes a procedure for configuring the SRG50 for CS 1000 Emergency Services Access.

Chapter 6
Setting up the private VoIP network

To provide SRG functionality and to take advantage of VoIP technology, a private VoIP network is required between the SRG50 and the main office. This chapter details the procedures for establishing appropriate WAN connections to enable a VoIP network between the main office and SRG50 branch locations. Before proceeding, ensure that IP networking from the SRG50 to the WAN, and from the main office call server to the WAN have been configured and tested (Figure 13). SRG-specific configuration establishes the VoIP network (Figure 14).

Figure 13 IP networking, SRG50 to WAN, main office to WAN

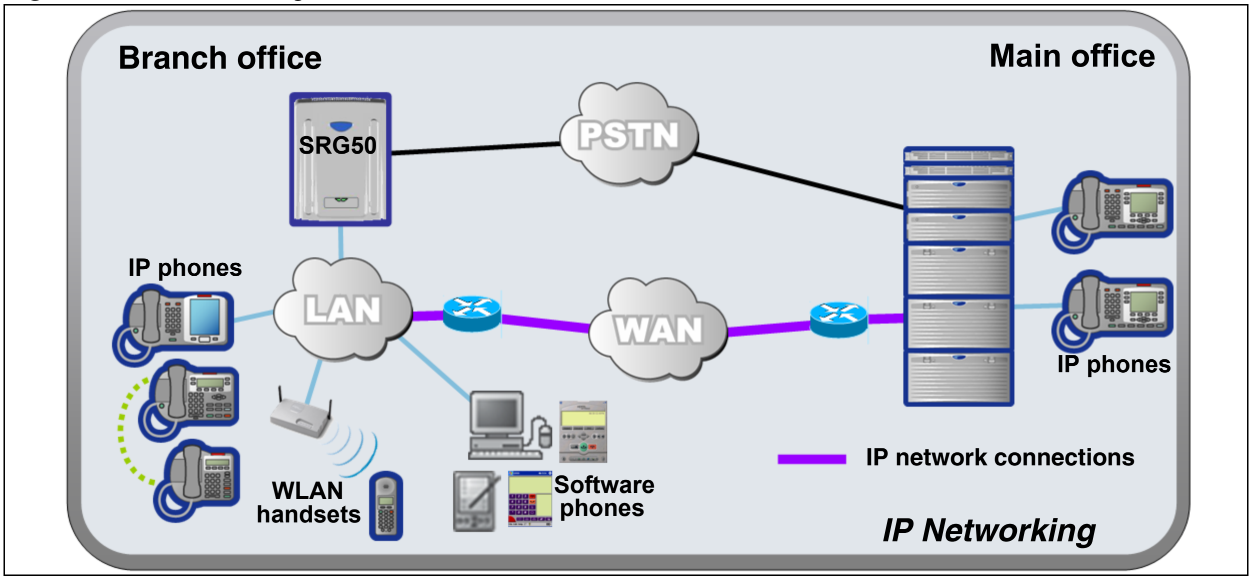
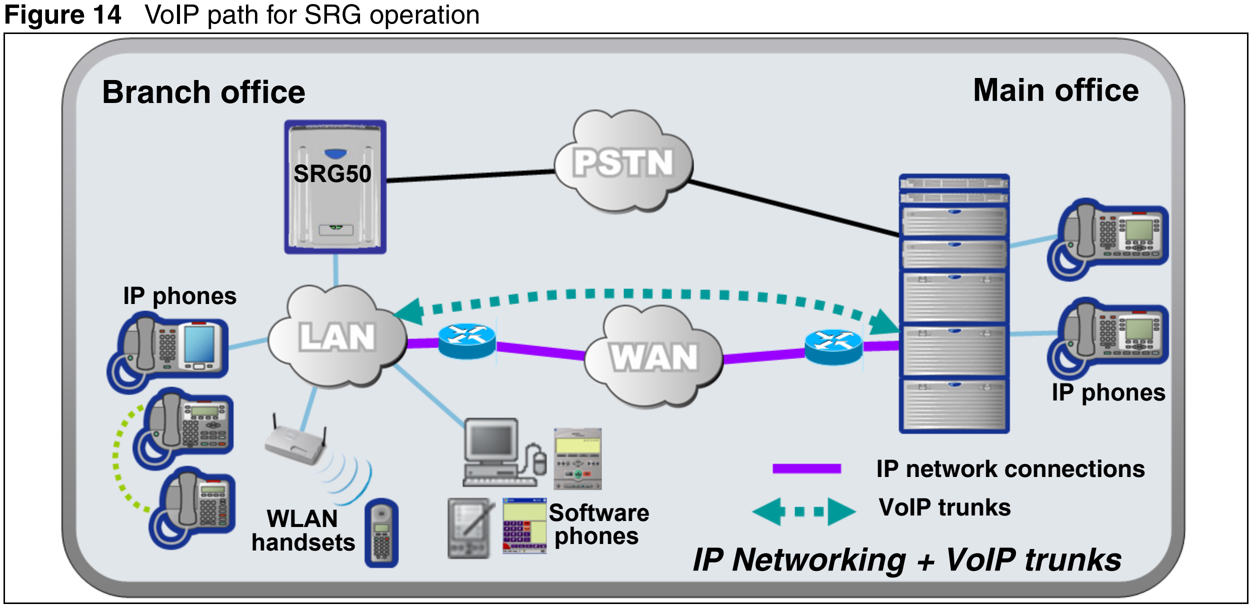


Figure 14 VoIP path for SRG operation



Generic procedures for setting up a private network on the SRG50 are covered in the *BCM50 Networking Configuration Guide* (N0027156). Items to address when establishing the private VoIP network between the SRG50 and the main office are:

- "Basic parameters" on page 77
- "Private dialing plan" on page 78
- "Meridian Customer Defined Network (MCDN)" on page 79
- "QoS settings (codec, jitter buffer, and related items)" on page 80
- "Network security" on page 82
- "Configuring VoIP trunking" on page 83
- "Line pools" on page 86
- "Call routing: introduction" on page 86
- "Call routing: configuring for outgoing calls" on page 86
- "Call routing: providing access to the SRG50 PSTN" on page 88
- "Main office information" on page 89
- "External attendant support" on page 90

Basic parameters

The following table provides a record of basic parameters that are significant for SRG operation. Typically, these parameters are specified as part of BCM50 foundation activities; in most cases, their configuration is not covered in the *Survivable Remote Gateway 50 Configuration Guide*.

Table 11 Basic parameters (Sheet 1 of 2)

Parameter	Value	Context
DN length		Configured as part of BCM50 foundation configuration. There are four DN lengths to consider. 1. **Configuration > Telephony > Dialing Plan > General > Dialing Plan - General panel > Global Settings** subpanel **> DN length (intercom)** field This is the internal DN length. That is, the length of DNs for calls between telephones on the SRG50. 2. **Configuration > Telephony > Dialing Plan > Public Network > Dialing Plan - Public Network** panel> **Public Network Settings** subpanel > **Public Received number length** field For calls originating from the PSTN, this establishes the number of digits the SRG50 retains. 3. **Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings** subpanel **> Private Received number length** field For calls originating from a private network, this establishes the number of digits the SRG50 retains. 4. **Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings** subpanel **> Private DN length** field Used for DPNSS applications only. Refer to the *BCM50 Networking Configuration Guide* (N0027156).
DN range		Configured as part of BCM50 foundation configuration. DN range of the SRG50 is set by the hardware configuration and keycodes. Actual numbering is contiguous from the Start DN: **Administration > Utilities > Reset > Reset** panel > **Cold Reset Telephony Services** button **> Cold Reset Telephony** pop-up window **> Start DN** field
Destination code for VoIP trunks		Configured for advanced routing. Refer to "Call routing: configuring for outgoing calls" on page 86 VoIP destination code = VOIP Trunk Access Code* = AC1** * VOIP Trunk Access Code is entered on the main office settings panel. Refer to the server-specific chapters for details. ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>.
SRG50 IP address		Configured as part of BCM50 foundation configuration. Path: **Configuration > System > IP Subsystem > IP Subsystem** panel **> General Settings** tab **> IP Settings details** subpanel

Table 11 Basic parameters (Sheet 2 of 2)

Parameter	Value	Context
SRG50 net mask		Configured as part of BCM50 foundation configuration. Path: **Configuration > System > IP Subsystem > IP Subsystem** panel **> General Settings** tab **> IP Settings details** subpanel
IP address of SRG50 gateway		Configured as part of BCM50 foundation configuration. Path: **Configuration > System > IP Subsystem > IP Subsystem** panel **> General Settings** tab **> IP Settings details** subpanel
VLAN		If the SRG50 operates as part of a VLAN, obtain the required identifiers from the VLAN administrator. **Configuration > Data Services > DHCP Server > DHCPServer** panel **> IP Terminal DHCP Options** tab **> VLAN Identifiers** subpanel
PSTN number for dialing into the main office from the SRG50 when in local mode		Required to specify a PSTN fallback route. Refer to "Call routing: configuring for outgoing calls" on page 86.
PSTN number for dialing into the SRG50 from the main office when in local mode		Required for main office configuration.

Private dialing plan

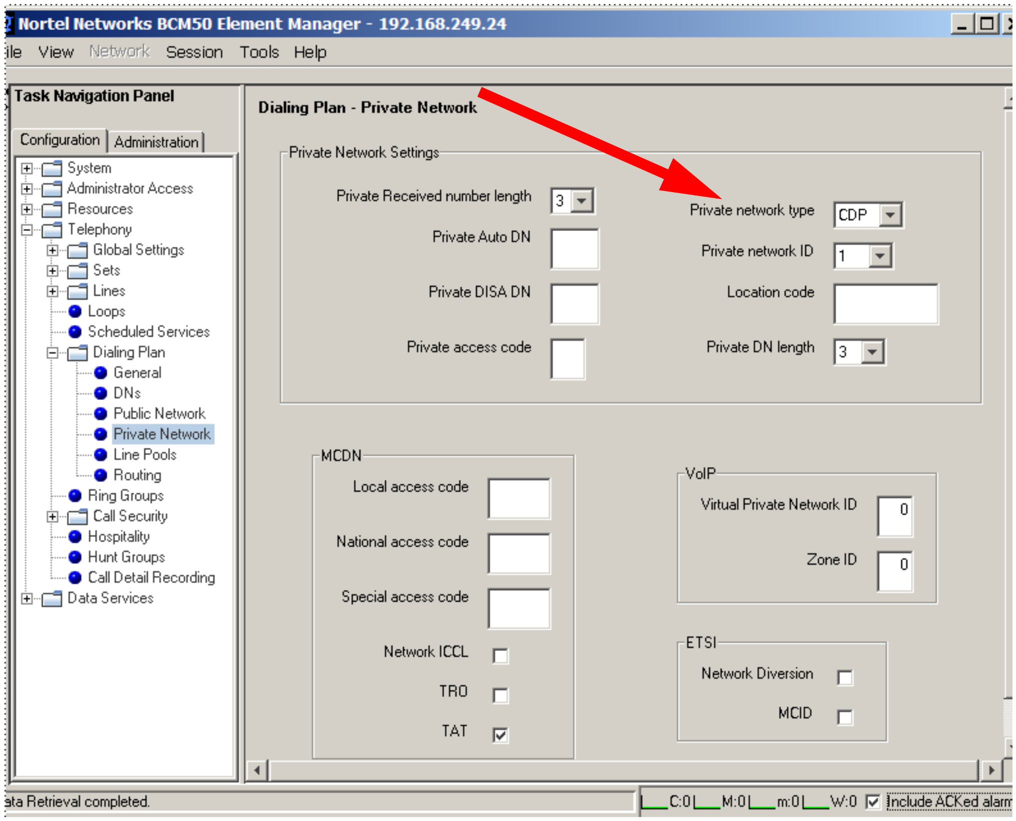
For SRG50 operation, either a coordinated dialing plan (CDP) or a uniform dialing plan (UDP) can be configured. Nortel recommends CDP because it requires the least dialing manipulation between the SRG50 and the main office call server. The dialing plan choice also determines whether the DN on the SRG50 matches the BUID (CS 1000) or DN (CS 2000) on the main office.

Dialing plans between the SRG50 and the main office call server must be compatible. Private dialing plan configuration is described in detail in the *BCM50 Networking Configuration Guide* (N0027156).

The type of dialing plan, CDP or UDP, is determined by the main office configuration.

The path to the SRG50 Element Manager panel for setting up the dialing plan is **Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network** panel **> Private Network Settings** subpanel (Figure 15).

Figure 15 Dialing Plan — Private Network panel, Private Network Settings subpanel



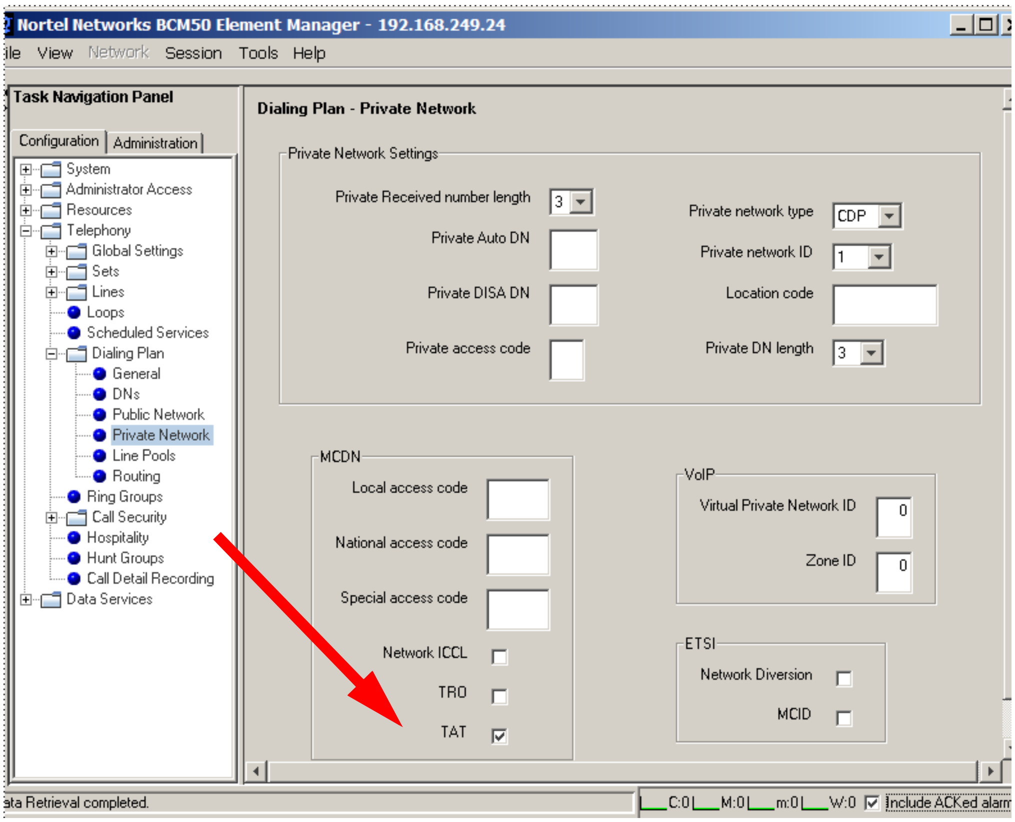
Meridian Customer Defined Network (MCDN)

MCDN is automatically activated when the system is converted to SRG operation. To ensure that redirected IP telephones can transfer calls to the SRG50 local telephones, trunk anti-tromboning (TAT) must be enabled.

To enable MCDN TAT

1 Use the SRG50 Element Manager to enable TAT by selecting the **TAT** checkbox at **Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network** panel **> MCDN** subpanel (Figure 16).

Figure 16 Dialing Plan — Private Network panel, MCDN subpanel



QoS settings (codec, jitter buffer, and related items)

Quality of Service (QoS) settings for the VoIP trunks at the SRG50 are determined by the main office settings; the SRG50 settings must match the main office. Use the following table to record the main office settings, to facilitate configuration, and to provide a record of the datafill.

Table 12 Main office QoS settings (Sheet 1 of 2)

Media parameter	Setting
Codec Preferences	
Silence compression (yes/no)	

Table 12 Main office QoS settings (Sheet 2 of 2)

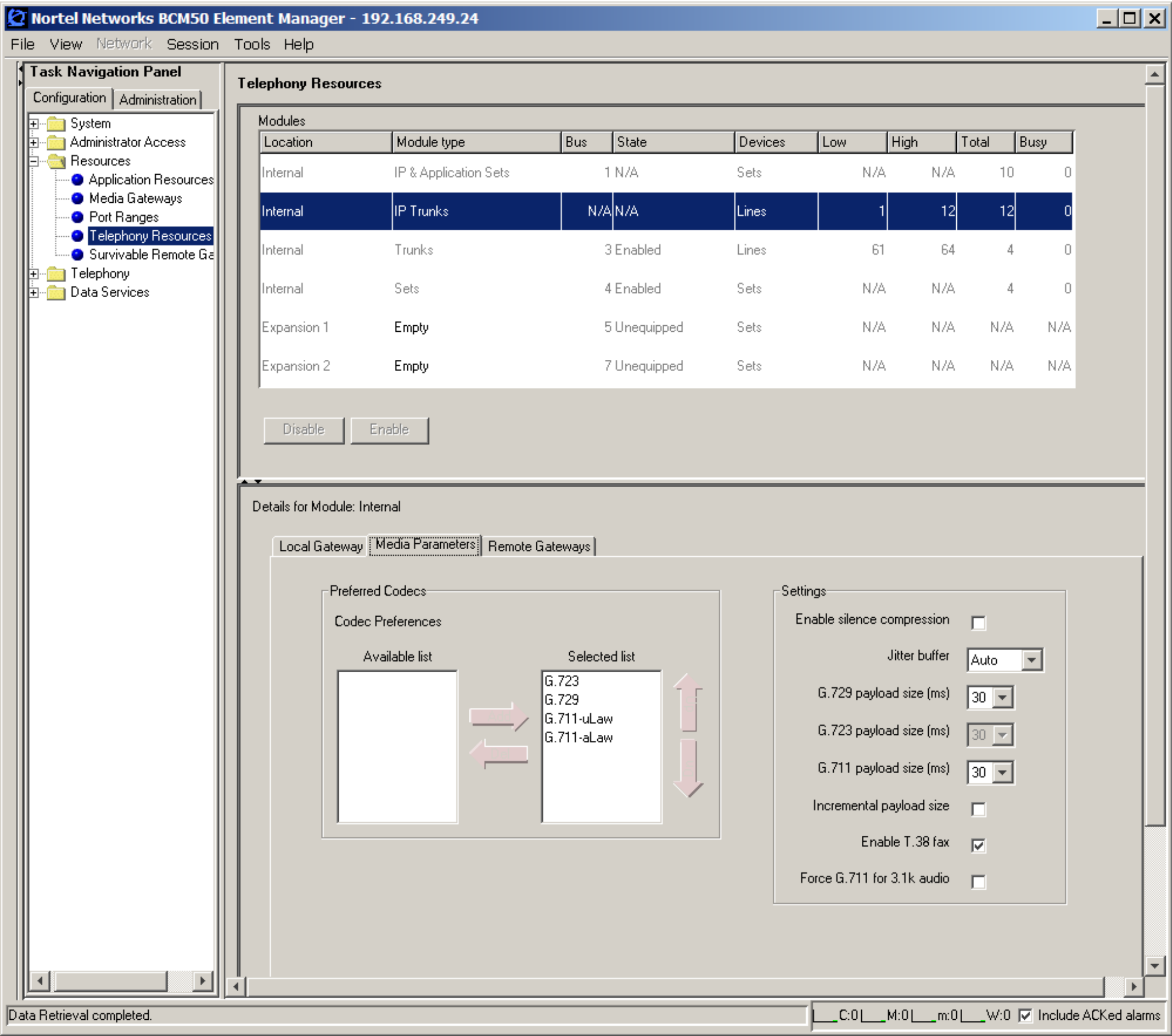
Media parameter	Setting
Jitter buffer	
G.729 payload size (ms)	
G.723 payload size (ms)	Supported for IP telephones for local calls only. Refer to "IP telephone codec and jitter settings" on page 65 for details.
G.711 payload size (ms)	
Incremental payload size (yes/no)	
T.38 fax (yes/no)	
Force G.711 for 3.1k audio (yes/no)	

On the SRG50 Element Manager, enter QoS settings for VoIP trunks through the **Telephony Resources** panel.

To enter the QoS settings for VoIP trunks

1. On the SRG50 Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
2. On the **Modules** panel, locate the **Module type** column and select the **IP Trunks** row (Figure 17).
3. On the **Details for Module: Internal** subpanel, select the **Media Parameters** tab.
4. Refer to the main office QoS settings recorded in the preceding table and enter the appropriate values.

Figure 17 Modules panel, IP Trunks



Network security

Firewall configuration for SRG50 is the same as for the BCM50 and is detailed in the *BCM50 Networking Configuration Guide* (N0027156). Firewalls cannot be configured to allow VoIP pass through. Instead, the SRG50 supports IPsec tunnels to provide VoIP pass through. IPsec tunnels are also covered in the *BCM50 Networking Configuration Guide* (N0027156).

Configuring VoIP trunking

When the SRG50 is operating in normal mode, connectivity to the main office call server is over VoIP trunks. An SRG50 can support up to 24 VoIP trunks.

Configuring VoIP trunks has three components:

- “Configuring fallback” on page 83
- “Gatekeeper routing (CS 1000 only)” on page 84
- “Line pools” on page 86

Configuring fallback

For SRG operation, fallback and gatekeeper configuration is common to all VoIP trunks.

To enable fallback

1 Access the IP Trunks configuration panel on the SRG50 Element Manager (**Configuration > Resources > Telephony Resources > Modules** panel **> Module type** column: select **IP Trunks** (Figure 18).

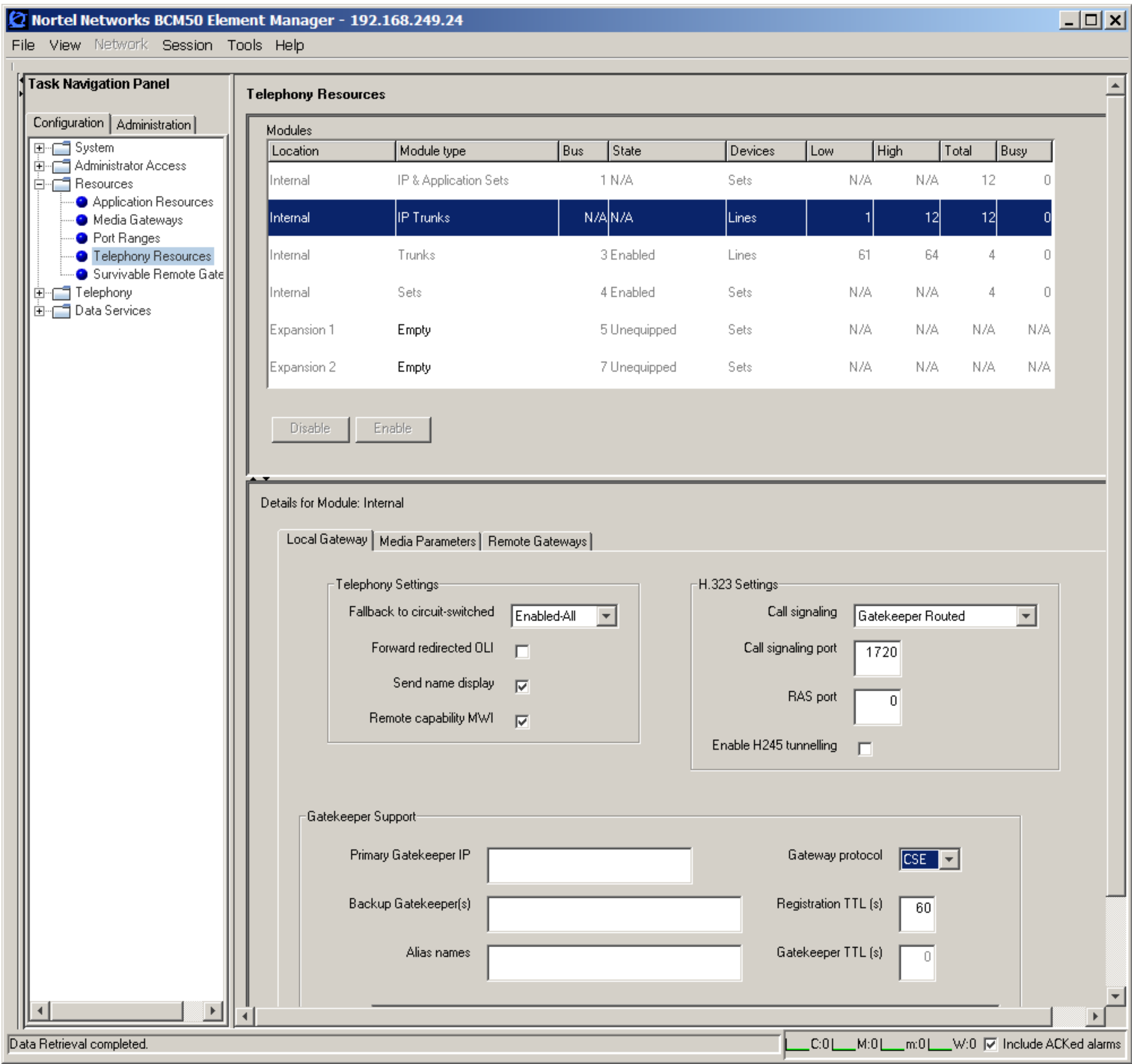
2 On the **Details for Module: Internal** details panel, select the **Local Gateway** tab.

3 On the **Telephony Settings** subpanel, locate the **Fallback to circuit-switched** field and use the pull-down menu to:

  a select **Enabled-All** if you want calls to be able to fallback to PSTN trunks if connectivity to the main office is lost.

  b select **Disabled** if fallback is not required.

Figure 18 IP Trunks panel, Local Gateway



Gatekeeper routing (CS 1000 only)

The gatekeeper routes the calls based on an internal numbering table. Ensure that the gatekeeper administrator has a list of the numbers that identify the SRG50 and the SRG50 PSTN.

Examples:

- If the system is running with a CDP dialing plan and the SRG50 DN range is from 3000 to 3199, the gatekeeper requires this information to establish that calls starting with 30 and 31 are routed to the SRG50.

- If the PSTN to which the SRG50 connects has a location code of 521, the gatekeeper must have a record of this code in the SRG50 list so that main office calls to the SRG50 PSTN can be routed correctly.

To configure gatekeeper settings

1. Access the IP Trunks configuration panel on the SRG50 Element Manager (**Configuration > Resources > Telephony Resources > Modules** panel **> Module type** column: select **IP Trunks** (Figure 18).
2. On the **Details for Module: Internal** details panel, select the **Local Gateway** tab.
3. On the **H.323 Settings** subpanel, locate the **Call signaling** field and use the pull-down menu to select **Gatekeeper Routed**.
4. On the **Gatekeeper Support** subpanel, locate:
   a. the **Primary Gatekeeper IP** field and enter the IP address for the **Primary Network Routing Services Address***
   b. the **Backup Gatekeeper(s)** field and enter the IP address for **Alternate Network Routing Services Address***
   c. the **Alias*** names field and enter the H.323 ID of the SRG50 (refer to the *BCM50 Networking Configuration Guide* (N0027156) for naming conventions)
   d. the **Gateway** protocol field and use the pull-down menu to select **CSE**.

   * See "CS 1000 information for the SRG50" on page 43.

Line pools

Both VoIP trunks and PSTN trunks must be configured in separate line pools. In the default state, all VoIP trunks are assigned to line pool BlocA and all PSTN trunks are assigned to line pool A. It is not necessary to reassign the line pools.

Instructions for configuring line pools is provided in the *BCM50 Networking Configuration Guide* (N0027156).

Call routing: introduction

Call routing is covered in depth in the *BCM50 Networking Configuration Guide* (N0027156). The instructions in the *Survivable Remote Gateway 50 Configuration Guide* are a cryptic abbreviation of that material, and only cover procedures that are specific to SRG50 operations; that is, for calls from redirected IP telephones. Refer to the *BCM50 Networking Configuration Guide* (N0027156) if more detailed information is required.

Note: CS 1000 only

On a CS 1000, the DNs for the main office telephones system are marked off by the vacant number routing feature. The SRG50 does not support Vacant Number Routing (VNR).

Instead, the SRG50 uses Call Forward All Calls to emulate VNR for the SRG50 IP telephones that are in normal mode. When the telephones switch to local mode, Call Forward All Calls is cancelled for those telephones.

A single destination code and route (or a group of destination codes and routes) can be set up on the SRG50 to route all the calls that are not terminated locally by the SRG50. These calls are routed over the VOIP trunks. In the case where the VoIP trunks become unavailable, the calls can be routed to the proper location using PSTN fallback. This is similar to the VNR feature in CS 1000.

Call routing: configuring for outgoing calls

To configure routing for outgoing calls

1 Create a schedule.

  a Access the **Scheduled Services** panel (**Configuration > Telephony > Scheduled Services**).

  b Select a **Schedule** (**Schedule 4**, for example).

  c Change the schedule name (optional).
  In this procedure, the name **SRG** is used as the name of the schedule.

- **d** Change the schedule time so that the schedule runs continuously (**Start Time** 00:00:00, **Stop Time** 23:59:59, MTWTFSS).
- **e** Change the routing service (**Routing Svc**) to **Auto**.
- **f** Select **Overflow**.

2 Define a route for calls to the main office over the VoIP trunks and a route to the main office over the PSTN.

- **a** Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab.
- **b** Add a new route (for example, 998).
- **c** Ensure that the **DN Type** is **Public (Unknown)**.
- **d** Assign the VoIP line pool to the route (select the line pool from the **Use Pool** pull-down menu; default is **BlocA**).
- **e** Add another new route (for example, 999).
- **f** Ensure that the **DN Type** is **Public (Unknown)**.
- **g** In the **External Number** field, enter the **PSTN number** of the main office.
- **h** Assign the PSTN line pool to the route (select the line pool from the **Use Pool** pull-down menu; default is **A**).

3 Add a destination code to provide access to the newly created routes. This code is used in both normal and local modes for dialing the main office from the SRG50 site.

- **a** Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab.
- **b** Add a new destination code (for example, **678**).
- **c** With the **Destination Code** row highlighted, select the **SRG** schedule from the **Alternate Routes** list (**Alternate Routes for Destination Code** details panel).
- **d** In the **First Route** field, enter **998** (the VoIP route).
- **e** In the adjacent **Absorbed Length** field, select the number of digits to be absorbed. The **Absorbed Length** applies to the digits of the destination code only.

  Typically, the **Absorbed Length** is **All** for UDP and **0** for CDP.
- **f** In the **Second Route** field, enter **999** (the PSTN route).
- **g** In the adjacent **Absorbed Length** field, select the number of digits to be absorbed.

Depending on the dialing plan, the destination code is integrated with the DN or is dialed as a prefix to the DN. When a user calls the main office, the SRG50 examines the destination code to determine the routing. If the **First Route**, the VoIP trunks, is unavailable, the call is routed to the **Second Route**, the PSTN, and the **External Number** is called. Because **Overflow** was selected, if both the **First Route** and the **Second Route** are unavailable, the call is routed using the **Normal Route** specified in the **Normal Route** column of the **Destination Codes** table. Because **Auto** was selected, the routing occurs without manual intervention.

Call routing: providing access to the SRG50 PSTN

Access to the SRG50 PSTN is required for:

- calls to the SRG50 PSTN from SRG50 telephones or redirected SRG50 IP telephones in local mode
- calls from the SRG50 PSTN to redirected SRG50 IP telephones in normal mode
- calls from main office telephones to the SRG50 PSTN, using the VoIP trunks

To achieve this access, a remote access package for the VoIP trunks and a destination code for the PSTN must be configured.

Remote Access Package for VoIP trunks

The SRG50 views all calls coming in over the VoIP trunks as remote access calls, even though the VoIP pathway is a dedicated trunk to another closed system.

To allow tandem dialing from the main office through the SRG50 PSTN, or to redirect SRG50 IP telephones to use the SRG50 local PSTN, you must specify a remote package that provides access to the PSTN line pool. This remote package is then assigned to each VoIP trunk.

To configure remote access packages

1. Set up a remote access package for the PSTN line pool (**Configuration > Telephony > Call Security > Remote Access Packages**).
2. Assign the package to each VoIP trunk (**Configuration > Telephony > Lines > Active VoIP Lines > All Lines** panel **> Trunk Type** column **>** select the desired trunk from the table **> Details for Line details** panel **> Restrictions** tab **> Use remote package** field).

Configuring PSTN destination codes

To allow SRG50 telephones to dial out over the PSTN and to allow main office telephones to tandem out through the local SRG50 PSTN, you need to define a destination code that accesses the PSTN line pool without an **External Number**. Frequently, this code is 9, but it does not have to be.

The following procedure provides a basic PSTN routing setup.

To configure destination codes for the PSTN

1. Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Routes** tab. Refer to the *BCM50 Networking Configuration Guide* (N0027156) to:
   a. Add a new route (for the PSTN line pool).
   b. Change the **DN Type** to **Public**.
   c. Leave the **External Number** field blank.

d Assign the PSTN line pool to the route (select the line pool from the **Use Pool** pull-down menu).

2 Access the **Dialing Plan - Routing** panel (**Configuration > Telephony > Dialing Plan > Routing**) and select the **Destination Codes** tab. Refer to the *BCM50 Networking Configuration Guide* (N0027156) to:

a Add the destination code to be used to access the local (SRG50) PSTN.

Users on both SRG50 and main office telephones dial this destination code to access the local (SRG50) PSTN. If this code goes only to the SRG50 PSTN, enter 9 + Wild Card 1. This wild card allows any numbers not used by other 9-based destination codes.

The default line pool access code for pool A is 9. Delete this access code before you attempt to create a destination code with 9.

In normal mode, the destination code is forwarded from the main office to the SRG50 for SRG50 IP telephone calls that are connecting to the SRG50 PSTN.

CS 1000 Considerations:

1. For main office programming, this code is the offnet dialing code that the gatekeeper recognizes for routing to the SRG50.
2. At the main office, zone-based digit manipulation is used to add a Zone Digit Prefix (ZDP) to PSTN calls from SRG50 IP telephones. The ZDP allows the main office to differentiate between local PSTN calls made from main office telephones (to the main office PSTN) and PSTN calls made from SRG50 IP telephones (to the SRG50 PSTN). The main office administrator for the CS 1000 supplies this ZDP with the prerequisite information.

b Assign the Normal and SRG scheduled route for the two destination codes.

Main office information

The SRG50 requires information about the main office call server that is not needed for a BCM50. The SRG50 Element Manager accommodates this information with SRG-specific panels that are activated after the SRG50 keycode is applied. The information required for these panels is specific to the main office call server. Refer to the server-specific chapters for details (“CS 1000 considerations” on page 33 and “CS 2000 considerations” on page 53).

External attendant support

The SRG50 can use the BCM50 Selective Line Redirection capability to provide an external attendant. If the attendant is located in the main office, there are two ways to maintain the attendant if the VoIP trunks become unavailable:

1. A fallback (or Prime) DN at the SRG50 can be specified. Since this DN is likely to receive calls in a WAN failure scenario, it must be an IP telephone that can transfer the calls to the desired party. If the IP set is also a redirected IP set, there is a period of time where inbound calls are un-routable, until the IP set falls back to the SRG50.
2. A fallback route to the main office call server over the PSTN can be configured. At the main office, vacant number handling (such as routing to voice mail) can be applied.

Note: The SRG50 does not have local attendant capability.

Chapter 7
Considerations for PSTN access and analog devices

SRG-specific items relevant to PSTN trunks and analog devices include:

- “Considerations for PSTN access” on page 91
- “Considerations for analog devices” on page 92

Considerations for PSTN access

1 PSTN access
To provide access to the SRG50 PSTN when the SRG50 is in local mode, or to be able to set up tandem dialing from the main office through the SRG50 to the SRG50 PSTN, one or more PSTN trunks must be configured on the system. Refer to the *BCM50 Networking Configuration Guide* (N0027156).

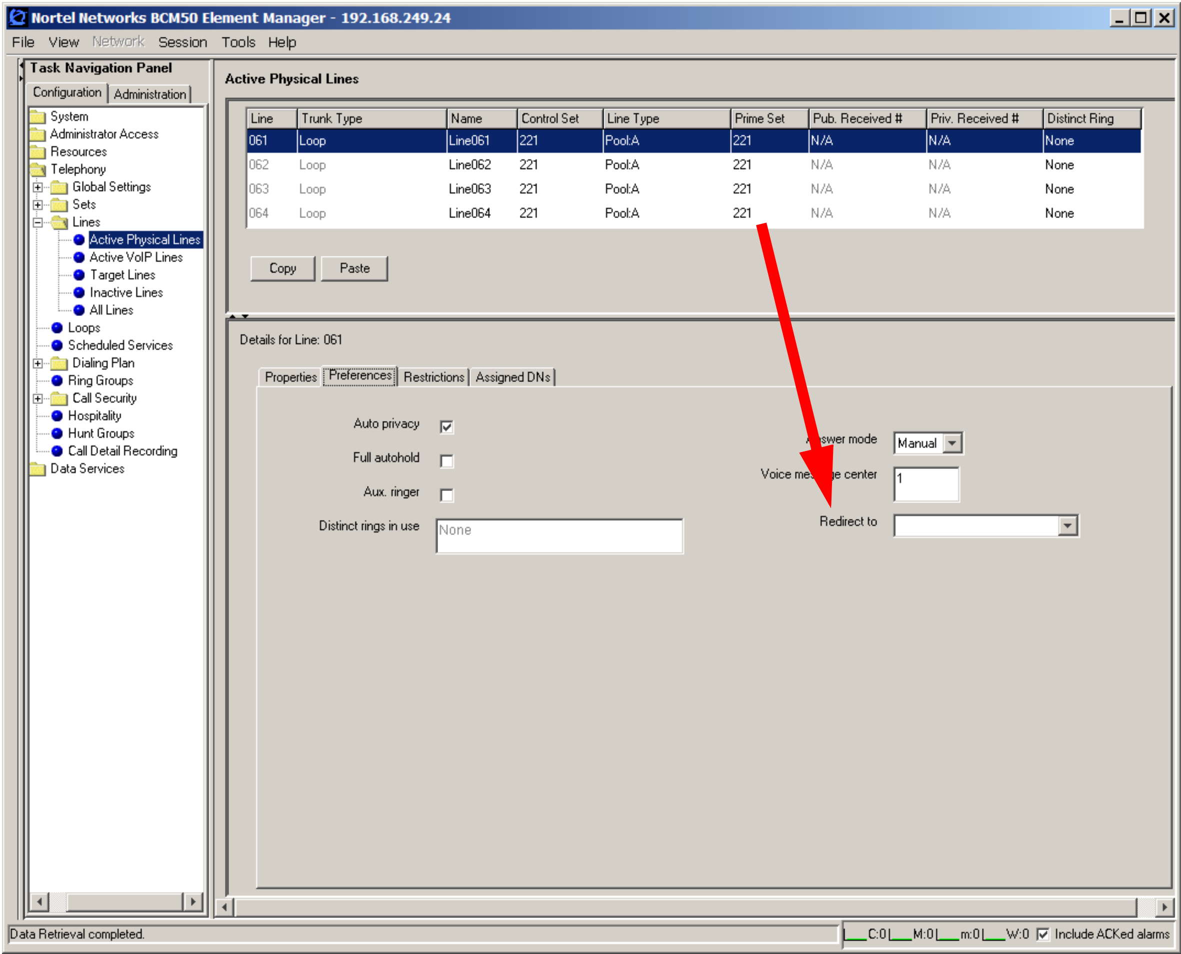
2 Tandem calls
The SRG50 considers all calls coming in over the VoIP trunks as remote access calls, even though the VoIP pathway is a dedicated trunk to another closed system.

To allow tandem dialing from the main office to the SRG50 PSTN, or to allow redirected SRG50 IP telephones to use the SRG50 local PSTN, a remote access package must be specified to provide access to the PSTN line pool. This procedure is covered in “Call routing: configuring for outgoing calls” on page 86.

3 Manual- and auto-answer lines
If the trunk is configured as a manual-answer line:
Enter the line pool access code and the dial string for the main office attendant telephone in the Redirect to field (Configuration > Telephony > Lines > Active Physical Lines).

If the line is an auto-answer line in normal mode, incoming call requests are automatically call forwarded to the main office. When the SRG50 IP telephones revert to local mode, the system discontinues Call Forward All Calls and calls are delivered directly to the SRG50 IP telephones at the SRG50.

Figure 19 Configuring manual and auto-answer lines for SRG operation



Considerations for analog devices

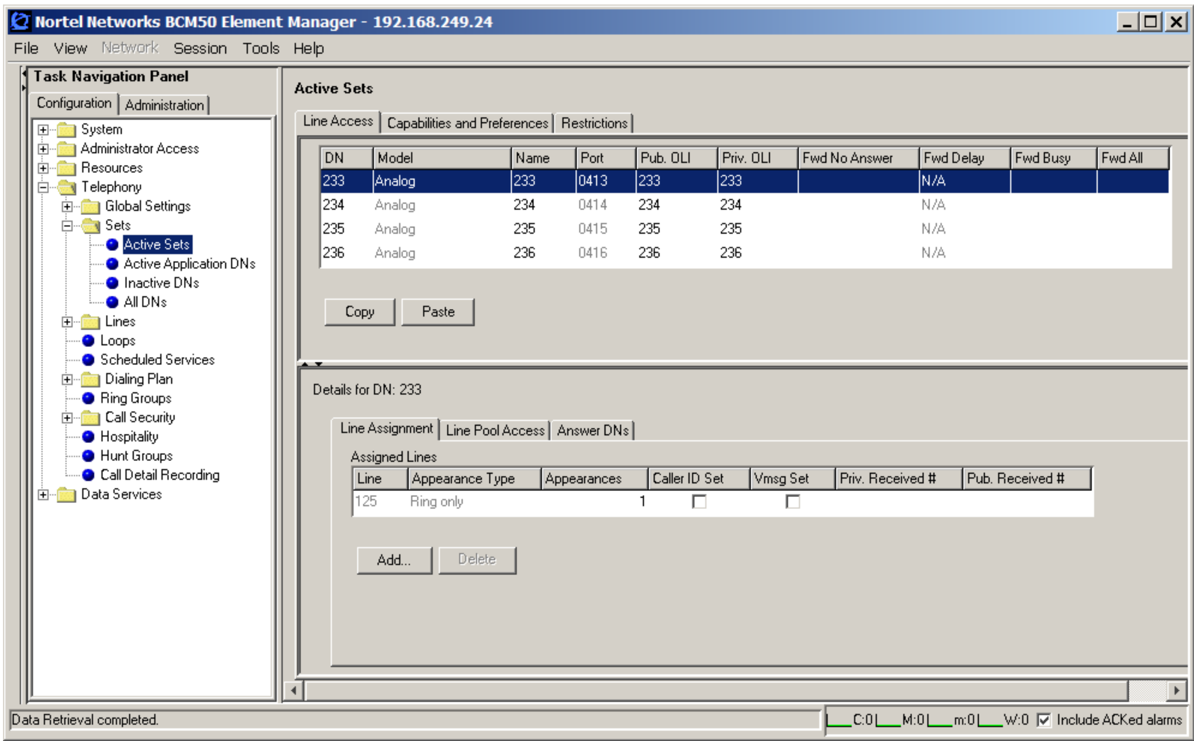
1 **Basic operation**
Analog telephones and devices connected to the SRG50 always function as local telephones to the SRG50. They can use the VoIP trunk to the main office using access codes or destination codes, if the VoIP trunk line pool is assigned to the device, but the main office does not have any settings or administration for these devices.

2 **Access to system features**
Analog telephones do not have a Feature key. Instead, they use a Link (*) key to access system features. If you leave the analog telephone records at the default settings, these telephones have greater feature access on the SRG50 than the IP telephones in local mode. If you do not want different feature access on the analog telephones, turn the unwanted settings off as you program the telephone.

To configure the DNs for analog devices

1 On the BCM50 Element Manager, navigate to the Active Sets panel (**Configuration > Telephony > Sets > Active Sets** (Figure 20), select the **Line Access** tab.

Figure 20 Active Sets panel



2 Identify the DNs for which the Model is Analog and align the settings for each with the information in the following steps.

Note: The Private OLI field (**Priv. OLI**) defaults to the DN. **Do not change this number.**

3 To support outgoing number display over the PSTN, enter the public access number for the telephone in the Public OLI field (**Pub. OLI**).

4 To assign specific PSTN lines to each telephone, add the line(s) (**Details for DN details panel > Line Assignment** tab). You would do this if, for example, you want one user to field all customer calls when the system is in local mode.

5 Ensure that the **Appearance Type** (**Details for DN details panel > Line Assignment** tab) is set to **Ring only**.

6 Assign the target line to the telephone (**Details for DN details panel > Line Pool Access** tab).

7 Assign both the PSTN and VoIP trunk line pools to all telephones that are allowed to make calls over the PSTN or to the main office over the VoIP trunk.

If you want the analog telephones to emulate local mode call functionality always, assign only the PSTN line pool to the analog devices.

8 On the **Active Sets** panel (**Configuration > Telephony > Sets > Active Sets** (Figure 20), select the **Capabilities and Preferences** tab.

9 Set **Handsfree** to **None**.

10 Clear the **HF Answerback** check box.

11 Clear the **Paging** check box.

12 Select the **Allow Redirect** check box if you want the user to be able to call forward to the main office or redirect lines to the main office.

Chapter 8
Troubleshooting

Potential problems, probable causes, and suggested solutions for SRG-specific configuration and operating troubles are categorized under the following topics:

IP telephone troubleshooting

Table 13 IP telephone troubleshooting (Sheet 1 of 2)

Issue / Problem	Probable Cause / Solution
Telephone does not connect to system	If an IP telephone does not display the text `Connecting to server` within two minutes after power up, the telephone was unable to establish communications with the SRG50. Double check the IP configuration of the telephone and the IP connectivity to the SRG50 (cables, switches, and so on).
Slow connection between the handset and the Business Communications Manager	If the connection between the IP client and the SRG50 is slow (ISDN, dialup modem), change the preferred codec for the telephone from G.711 to G.729.
Block individual IP sets from dialing outside the system.	If you want to block one or more IP telephones from calling outside the system, use Restriction filters and assign them to the telephones you want to block. Restriction filters are set up under Configuration > Telephony > Call Security > Restriction Filters.

Table 13 IP telephone troubleshooting (Sheet 2 of 2)

Issue / Problem	Probable Cause / Solution
One-way or no speech paths	Signaling between the IP telephones and the SRG50 uses UDP port 7300. Voice packets are exchanged using the default RTP: • Source port (output filters)/Destination port (input filters): 28000 through 28511 for the VoIP gateway. Output filter Destination IP is set to ALL. Input filter Destination IP is the IP address of the SRG50 local gateway. • Source port (output filters)/Destination port (input filters): 5200 - 5201 for the IP telephones. Output Destination IP is set to ALL. Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Normal mode. • Source port (output filters)/Destination port (input filters): 51000 - 51184 for the local mode IP sets. Destination port (output filter) and Source Port (input filter) are set to ALL. Output Destination IP is set to ALL. Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Local mode. UniSTIM signals use specific source and destination ports: • Output filters: Source port, 5000; Destination port, 4100, 5100, 7300. Output filter Destination IP is the IP address of the main office TPS. • Input filters: Source port, 4100, 5100, 7300; Destination port, 5000.Input filter Destination IPs are the IP range for all IP telephones (behind the firewall) in Normal mode. If these ports are blocked by the firewall or NAT, you will experience one-way or no-way speech paths. **Firewall note:** If the firewall filter is set to **Pass Outgoing and Block Incoming Except IP Phones**, this allows only IP telephony registration traffic through, but blocks all other traffic, including H.323 calls on this interface. You must still specify an H.323 rule to allow IP call voice traffic.

IP terminal details

The following table summarizes the events that can be raised by the SRG50. The events and details appear in the SRG50 Element Manager at **Telephony > Sets > Active Sets > IP Terminal Details**.

Periodic retries may result in the same condition being detected over and over again. In these cases the SRG50 state machine uses flags to indicate that a given event has been logged.

Table 14 IP Terminal Details (Sheet 1 of 2)

Details	Event Id	Severity	Call Server Type	Comments
SRG Started	57000	Warning	All	Indicates that the SRG50 process has started.
SRG Shutdown	57001	Warning	All	Indicates that the SRG50 process has shut down.
DN:XXX, Test Local Mode	57002	Warning	All	Test Feature

Table 14 IP Terminal Details (Sheet 2 of 2)

Details	Event Id	Severity	Call Server Type	Comments
DN:XXX, Local Mode - Firmware is out of sync with Main Office Call Server.	57003	Warning	S1000	Indicates that IP set FW on main office has been upgraded and the required FW version is available on the SRG
DN:XXX, Local Mode - Set Firmware Upgrade in Progress	57004	Warning	S1000	The firmware required by the main office is being upgraded to the set,
DN:XXX, Normal Mode - Set Redirected to Main Office	57005	Warning	All	The set has been redirected to the main office.
DN:XXX, Local Mode - Redirection Pending (Set on call)	57006	Warning	All	The redirection of the set is pending as the set is on a call.
DN:XXX, Local Mode - Firmware Upgrade Pending (Set on call)	57007	Warning	S1000	The firmware upgrade to the set is pending as the set is on a call.
DN:XXX, Local Mode - Main Office Parameters Not Provisioned.	57008	Warning	All	The set is not provisioned to be redirected.
DN:XXX, Invalid ID (1) - No endpoint in Gatekeeper database	57250	Minor	S1000	Indicates configuration problem.
DN:XXX, Invalid ID (2) - ID unknown within the Call Server	57251	Minor	S1000	Indicates configuration problem.
DN:XXX, Invalid ID (3) - Endpoint in Gatekeeper database is Originating Call Server	57252	Minor	S1000	Indicates configuration problem.
DN:XXX, Local Mode - Net Connect Server Unreachable	57253	Major	S1000	Indicates either a configuration error, or a network connectivity error or the Net connect server has stopped.
DN:XXX, Local Mode - Main Office TPS Unreachable	57500	Major	All	Indicates either a configuration error, or a network connectivity error, or the MO TPS has stopped.
DN:XXX, Local Mode - Firmware is not available on the SRG	57501	Major	S1000	Indicates firmware required by the main office is not available in the SRG50.
SRG terminated unexpectedly.	57750	Critical	All	Indicates that a critical error caused the SRG50 process to terminate.

Probable causes for redirection failure

The IP telephone registration to the main office call server can fail due to improper configuration or lack of WAN connectivity. When a registration failure occurs, the error code and description is shown in the status field for the IP telephone in the IP Terminal Details field (refer to "IP terminal details" on page 96); the IP telephone remains registered with the SRG50 in local mode operation.

Definitive causes for registration failure depend on the main office call server.

For the CS 1000, causes can include:

- The registration password entered at the SRG50 does not match the installer password at the main office.
- The main office is unreachable.
- There is no endpoint configured for the user id or branch user id / TN combination.
- There is set firmware incompatibility between the remote and local offices.
- The actual IP telephone set type at the SRG50 does not match MOTN set type at the main office.
- The user id is registered and not idle.
- The user id entry in the gatekeeper database points back to the originating node.

For the CS 2000, there are fewer main office parameters to configure on the SRG50. As a result, there are fewer places where configuration may be an issue, but this also means that the difference between a WAN problem versus an address configuration problem is not as evident.

Troubleshooting fallback to local mode

If the system reverts to local mode and the problem is not the WAN link to the main office, check for:

a IP telephone firmware discrepancies

The SRG50 supports automatic firmware updates. These are covered in the server-specific chapters ("CS 1000 considerations" on page 33 and "CS 2000 considerations" on page 53). However, the possibility exists that a non-network reversion to local mode is caused when the IP telephone firmware has been updated on the main office and not on the SRG50.

Check the **IP Terminal Details** (refer to "IP terminal details" on page 96) for this **Status**:

```
Firmware is Out of Sync with the Main Office Call Server
```

The preferred way of handling firmware upgrades is to install the patch onto the SRG50 first, then on the main office equipment.

When the IP telephone firmware is updated on the main office, the main office redirects all SRG50 IP telephones back to the SRG50 for a firmware upgrade. If the SRG50 has already been patched with the new firmware, the telephone is upgraded when it registers with the SRG50. Once the telephone has the new firmware, the system automatically allows the telephone to reregister with the main office. If the correct firmware cannot be applied, for example because the SRG50 has not been upgraded with the new firmware, the telephone is redirected back to the main office.

b Gatekeeper failure (CS 1000 only)

If an IP telephone fails to establish communication with the gatekeeper when it tries to register to the main office, the telephone remains registered to the SRG50 and stays in local mode.

Troubleshoot the problem by checking the settings made when implementing the "CS 1000 information for the SRG50" on page 43 and "Configuring fallback" on page 83. If you make changes, manually redirect the telephones ("Manually redirecting IP telephones", below).

Manually redirecting IP telephones

To manually redirect an IP telephone to the main office

1. Access the **Telephony** > **Sets** > **Active** Sets > **IP Terminal Details** in the SRG50 Element Manager interface.
2. Click on the telephone listing that you want to redirect to normal mode.
3. Click the **Status** tab to view the **Status** field. If Status displays **Up,** the conversion was successful.

   a If the IP terminal does not register correctly with the main office, refer back to the IP Terminal Status tab, **Status** field and review the message to determine where the problem occurred. Refer to "IP terminal details" on page 96.

   b If the conversion occurred correctly, perform basic telephony tests to ensure that the telephones are working as expected:

   — Make and receive calls.

   — Check feature access.

   — Check voice mail access

   Refer to the feature guides for the main office application for specific information about making calls and using features.

Appendix A
Telephone features in normal and local mode

The information provided here is designed for distribution to telephone users at the SRG50.

Normal mode

In normal mode, IP telephones have the feature set that is supported by the main office. User cards are not supplied with the SRG50 because the feature set depends on the main office applications. If required, obtain user cards from the main office for normal mode features.

Features available to analog and ISDN telephones are provided by the SRG50 and depend on the SRG50 applications. Consult the SRG50 system administrator for a complete description.

A quick reference list to the default SRG50 features for an analog telephone are provided in “ATA extension features” on page 104. Consult the SRG50 administrator to determine if these features have been changed.

Local mode

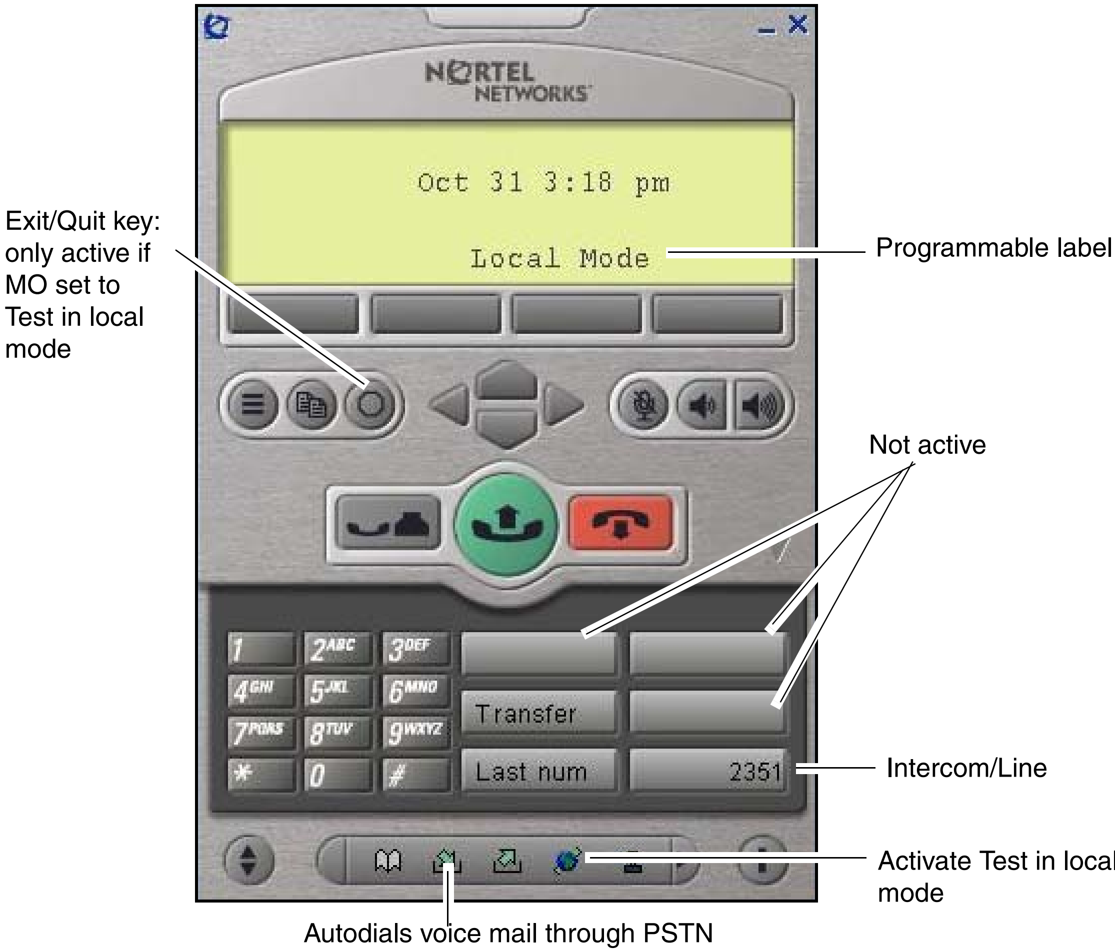
In local mode, call control and features are provided by the SRG50 are processed by the SRG50. Access to the main office is over PSTN lines; main office telephony features and applications are not available.

If routing and destination codes are set up as suggested in “Call routing: introduction” on page 86, the dialing sequence for the main office is the same as in normal mode.

For illustrations that show the default display settings for each type of IP telephone when the phone is in local mode, refer to:

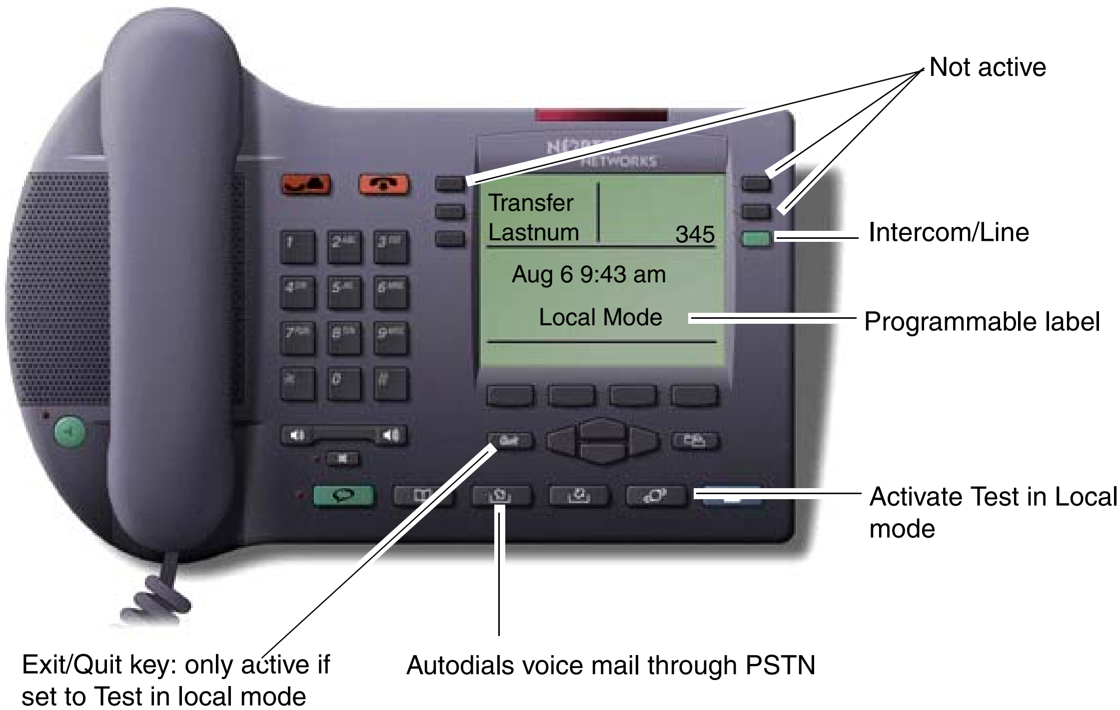
- “2050 Software Phone in Local Mode*” on page 102
- “2004 in Local mode” on page 103
- “2002 in Local mode” on page 104

2050 Software Phone in Local Mode*

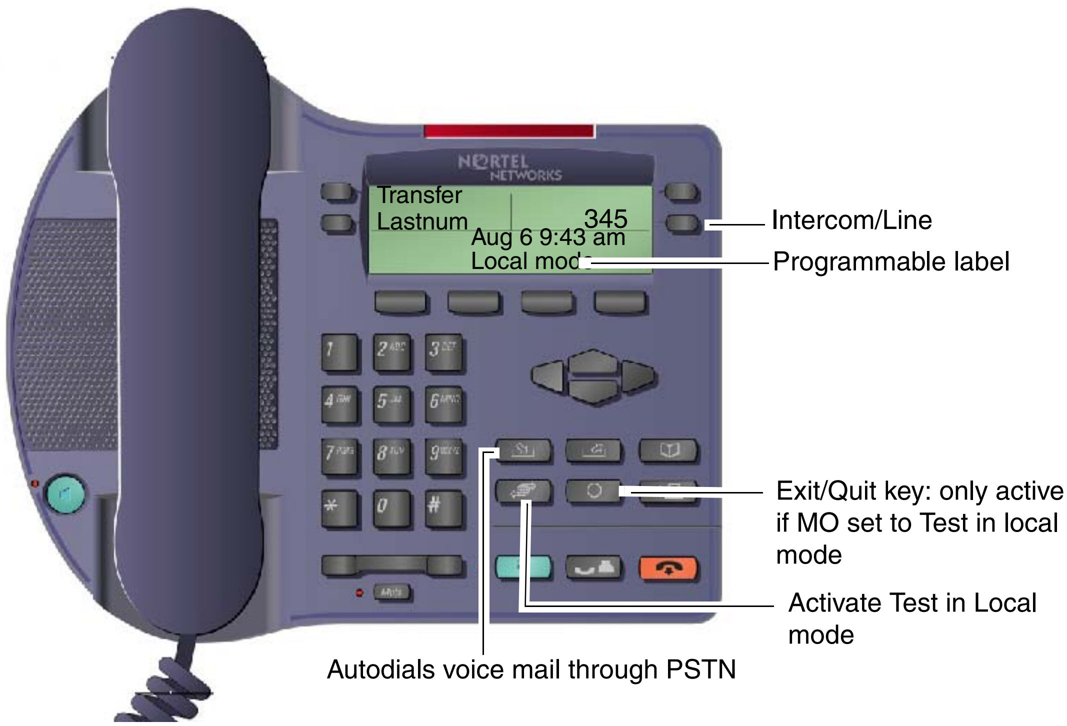


Note: MVC2050 and MVC2050E Softphones are supported on CS 1000 only.

2004 in Local mode



2002 in Local mode



ATA extension features

Analog telephones can be connected to the system through analog station modules or by installing an Analog Terminal Adapter (ATA) between the telephone and a digital station module. These telephones have only basic button configurations, so instead of using the feature key, press the link key (*) to invoke features on the system. Refer to the following list for the specific key sequence.

Feature	Activate	Cancel
Alternate line	LINK 2	
Call Forward (local system)	LINK *4	LINK #4
Call Forward (external system)	LINK *4 <dialed #> LINK 2	LINK #4
Call parking	LINK *74	
Call pick-up (Directed)	LINK *76	
Call pick-up (Group)	LINK *75	
Call Queuing	LINK *801	
Camp-on	LINK *82	
Conference call	LINK *3	
Hold Call (Exclusive)	LINK *79	
Hold Call (Public)	LINK 2	
Last Number Redial	LINK *5	

Feature	Activate	Cancel
Privacy control	LINK *83	
Link	LINK *71	
Pause	LINK *78	
Timed release	LINK *72	
Ring Again	LINK *2	LINK #2
Saved Number Redial	LINK *67	
Tones	LINK *809	LINK #809
Transfer	LINK *70	
Trunk Answer	LINK *800	
Voice Call	LINK *66	
Voice messaging - Internal		
Access mailbox	LINK *981	

Feature	Activate	Cancel		Feature	Activate	Cancel
Page - Intercom	LINK *61 and zone (0 to 6)			Leave a message	LINK *980	
Page - External	LINK *62					
Page - All	LINK *63 and zone (0 to 6)					

Glossary

BDP
: Both Dialing Plans.

  A dialing plan option that is supported on the main office only. The SRG50 supports CDP or UDP only. If the main office is running BDP, the SRG50 zone must be configured to allow either CDP or UDP, not both.

branch office
: A system that is remote from the main office but provides telephony services using the main office servers. When a branch office is a survivable remote gateway, telephony services are provided by the branch office if communication with the main office is lost.

call routing
: Coding that is configured on a system to ensure that outgoing calls are directed to the correct trunks and incoming calls are directed to the correct device(s) on the system. (see also: dialing plan)

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, consult the main office documentation that covers dialing plans.

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

FXO
: Foreign eXchange Office: an interface that connects to the PSTN central office and is the interface offered on a standard telephone. Example: RJ-11 connector that allows analog connection to the central office.

gatekeeper
: The gatekeeper is an IP network application that directs IP traffic to all the systems on the network. Parameters for both the main office and the SRG50 must be assigned to all gatekeepers on the network. If the gatekeeper is down, the SRG50 attempts to connect to the alternate gatekeeper, if there is one. If the alternate gatekeeper is also down, or there is no alternate gatekeeper, the SRG50 IP telephones remain registered with the main office, but calls cannot be sent to the SRG50.

gateway
: The IP portal on each system that establishes the VoIP trunk.

H.323
: An IP gateway protocol used by both the main office and the SRG50 to create VoIP trunking connections.

IP
: Internet Protocol

  IP specifies the format of packets, also called datagrams, and the addressing scheme in the TCP/IP protocol suite. Where IP defines the packet and addressing scheme, Transport Control Protocol (TCP) establishes a virtual connection between a destination and a source.

IP telephones
: Telephones that can connect directly with a TCP/IP network. Also known as internet telephones.

local mode
: The operating mode of redirected SRG50 IP telephones when connectivity with the main office is unavailable.

main office, main office call server
: The system that provides telephony services to redirected SRG50 IP telephones in normal mode.

NCS
Network Connection Server

The NCS is an H.323 gatekeeper. It provides standard H.323 gatekeeper functionality, as well as support for branch office and virtual office features.

normal mode
The operating mode of the SRG50 when connectivity with the main office is established.

QoS
Quality of Service

In IP telephony, QoS refers to the quality of the voice communication channel. There are inherent difficulties associated with transmitting voice over IP and quality of service is a significant challenge for service providers. QoS specifications allow service providers and their customers to establish and monitor acceptable levels of service.

steering codes
Steering codes are similar to line pool access codes and destination codes. Steering codes determine where a call gets routed.

TPS
(Internet Telephone) Terminal Proxy Server

A TPS controls the connection between IP telephones.

UDP
Uniform Dialing Plan

Each location within the network is assigned a Location Code. On a private network, this code precedes the directory number of the telephone being dialed. Depending on routing configuration, this number may be part of the destination code, or it may automatically be dialed out when the appropriate destination code is dialed before the directory number. Under the Uniform Dialing Plan (UDP), the SRG50 must include this code in the BUID.

UDP
User Datagram Protocol

A member of the TCP/IP protocol suite that transports data as a connectionless protocol, using packet switching. Generally, ports on the SRG50 support UDP.

VoIP trunk
Voice over IP trunk

A pathway between two systems that allows voice to be transmitted over an IP connection.

WAN
Wide Area Network

A computer network that spans a relatively large geographical area. The largest WAN in existence is the Internet.

ZDP
Zone Digit Prefix

The main office appends this number to an SRG50 local-PSTN call dialed from an SRG50 IP telephone. The number differentiates the call from a main office local-PSTN call dialed by a main office telephone.

Index

U

V

W