SRG50 5.0 Configuration Guide

BCM 5.0

Survivable Remote Gateway

Document Status: **Standard**

Document Number: **NN40140-500**

Document Version: **03.01**

Date: **August 2009**

Task List

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Getting started

About this guide

The *SRG50 5.0 Configuration Guide* describes how to install, configure, and maintain the Survivable Remote Gateway (SRG) 50 Release 5.0.

The SRG50 Release 5.0 is positioned as a cost-effective Small IP Branch Office solution for CS 1000 Main office systems. The SRG50 offers business continuity and public switched telephone network (PSTN) failover for voice over IP (VoIP) networks. An SRG provides transparent operation, feature and application 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. For more information on configuring the SRG50 for a CS1000 Main office, see *CS1000 Main Office Configuration Guide*.

The SRG50 Release 5.0 introduces SIP Trunking, extends Nortel IP Phone support and expands the number of survivable IP users from 32 to 80 with a single SRG application authorization code. It is provided as a cost-effective VoIP business continuity solution for small branch offices.

Audience

The *SRG50 5.0 Configuration 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.

Acronyms

The following is a list of acronyms used in this guide.

Table 1 Acronyms used in this guide (Sheet 1 of 3)

Acronym	Description
ACR	Alternative call routing
ANBWM	Adaptive network bandwidth management
ASM	Analog station module

Table 1 Acronyms used in this guide (Sheet 2 of 3)

Acronym	Description
ATA	Analog terminal adapter
BARS	Basic alternate route selection
BUID	Branch user ID
CDP	Coordinated dialing plan
DDIM	Digital drop and insert mux
DN	Directory number
DSC	Distant steering codes
DSM	Digital station module
DTM	Digital trunk module
ESA	Emergency services access
ESDN	Emergency services DN
FCAPS	Acronym for Fault, Configuration, Accounting, Performance, Security
FTP	File Transfer Protocol
FRL	Facility restriction level
GATM	Global analog trunk module
KEM	Key expansion module
KRS	Keycode retrieval system
LAN	Local area network
LSC	Local steering codes
MCDN	Meridian customer defined network
MOTN	Main office terminal number
MVC	Mobile voice client
NARS	Network alternate route selection
NBWM	Network bandwidth management
PSTN	Public switched telephone network
QoS	Quality of service
SFTP	Secure File Transfer Protocol
SPN	Special number
SRG	Survivable remote gateway
TAT	Trunk anti-tromboning
TSC	Trunk steering codes
UDP	Uniform dialing plan
VNR	Vacant number routing
VoIP	Voice over internet protocol
VPN	Virtual private network
VPNI	Virtual private network ID

Table 1 Acronyms used in this guide (Sheet 3 of 3)

Acronym	Description
WAN	Wide area network
ZDP	Zone digit prefix

Symbols and conventions used in this guide

These symbols highlight critical information for the SRG system.

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

Danger: Alerts you to conditions where you can get an electrical shock.

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

Note: Alerts you to important information.

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

Security Note: Indicates a point of system security where you can change a default or where the administrator must decide on the level of security required for the system.

Warning: Alerts you to ground yourself with an antistatic grounding strap before performing the maintenance procedure.

Warning: Alerts you to remove the main unit and expansion unit power cords from the AC outlet before performing any maintenance procedure.

These conventions and symbols represent the Business Series Terminal display and dialpad.

Convention	Example	Used for
Word in a special font (shown in the top line of the display)	`Pswd:`	Command line prompts on display telephones.

Convention	Example	Used for
Underlined word in capital letters (shown in the bottom line of a two-line display telephone)	PLAY	Display options on two-line display telephones. Press the button directly below the option on the display to proceed.
Dialpad buttons		Buttons you press on the dialpad to select a particular option.

These text conventions are used in this guide to indicate the information described.

Convention	Description
bold Courier text	Indicates command names, options, and text that you must enter. Example: Use the **info** 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 corresponding icon on the set you are using.

Related publications

This section provides a list of additional documents referred to in this guide.

Nortel BCM50 5.0 Installation Checklist and Quick Start Guide (NN40170-308)

Nortel BCM50 5.0 Installation and Maintenance Guide (NN40170-305)

Nortel Business Communications Manager 5.0 Configuration—System (NN40170-501)

Nortel Business Communications Manager 5.0 Administration and Security (NN40170-603)

Nortel Business Communications Manager 5.0 Configuration—Telephony (NN40170-502)

Nortel Business Manager 5.0 Configuration—Devices (NN40170-500)

Nortel Business Communications Manager 5.0 Installation—Devices (NN40170-304)

Nortel Business Communications Manager 5.0 Planning and Engineering (NN40170-200)

Keycode Installation Guide (NN40010-301)

Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50 (NN43001-307)

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 way to get technical support for Nortel products is from the Nortel Technical Support Web site:

www.nortel.com/support

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. More specifically, the site enables you to:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

Getting Help over the phone from a Nortel Solutions Center

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

In North America, call 1-800-4NORTEL (1-800-466-7835).

Outside North America, go to the following Web site to obtain the phone number for your region:

www.nortel.com/callus

Getting Help from a specialist by using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to:

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, contact the technical support staff for that distributor or reseller.

Chapter 2
SRG50 overview

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 CS1000 IP branch office environment.

SRG50 creation

An SRG50 is created by applying the appropriate SRG keycode to a functional BCM50 system. SRG50 is only supported on the BCM50 and BCM50b main units. Integrated router versions of the BCM50 (BCM50a, BCM50e, BCM50ba, and BCM50be) do not support the SRG50 application.

The *Nortel Business Communications Manager 50 5.0 Installation Checklist and Quick Start Guide* is provided on the SRG50 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 for your reference in the table BCM50 default IP addresses on page 15 and the table BCM50 default user names and passwords on page 15.

Table 2 BCM50 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 the OAM port and assigns the following IP address: **10.10.11.2**		

Table 3 BCM50 default user names and passwords

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

SRG50 keycode activation

To create an SRG50, use Nortel Business Element Manager to activate the SRG keycode on a BCM50 system (BCM50 or BCM50b main unit).

To activate the SRG keycode

1. Locate the SRG authorization codes supplied with your product.
2. Open Business Element Manager (see the *Nortel Business Communications Manager 5.0 5.0 Installation Checklist and Quick Start Guide*).
3. With Business Element Manager, connect to the BCM system that you want to convert to an SRG (see the *Nortel Business Communications Manager 5.0 5.0 Installation Checklist and Quick Start Guide*).
4. Navigate to the **Keycodes** panel (Configuration > System > Keycodes).
5. Click **Connect to Nortel Keycode Retrieval system** to obtain the keycode file for your system from the Nortel Keycode Retrieval System (KRS).

   For more information about keycodes, see the *Keycode Installation Guide*.
6. In the KRS, generate the keycode file for your system and save it on your management computer.

   Make sure the SRG feature is included in your keycode as well as any other features you require for your system.
7. In Business Element Manager, return to the keycodes panel.
8. Click **Load File**.
9. Browse to the location on your management computer containing the keycode file for this system.
10. Select the keycode file, and then click **Open**.

    The keycode file is applied.
11. Reboot your system to complete the creation of the SRG. Wait 2-3 minutes before rebooting after loading the keycode file.

To reboot the system

1. In Business Element Manager, navigate to the **Reboot** panel (Administration > Utilities > Reboot).
2. Click **Reboot**.

To verify that the SRG has been successfully created

1. In Business Element Manager, navigate to the **Keycodes** panel (Configuration > System > Keycodes).

   In the Feature licenses table, verify that the status of the SRG keycode is ACTIVE.
2. Open the **Resources** folder (Configuration > Resources).

Verify that there is a Survivable Remote Gateway panel.

SRG50 and BCM50 features comparison

The table Comparison of BCM50 and SRG50 on page 17 compares SRG50 and BCM50 features.

Table 4 Comparison of BCM50 and SRG50

Item	BCM50	SRG50
MBMs	See the *Installation and Maintenance Guide*	Recommended: ASM8+ (8 port Analog Station Module); DTM (Digital Trunk Module - 24 lines on either T1 or E1 or PRI); BRI4 (4 line BRI S/T Module); GATM4 (Global Analog Trunk MBM - 4 port); GATM8 (Global Analog Trunk MBM - 8 port): ADID4 (Analog Direct Inward Dial 4 Port): ADID8 (Analog Direct Inward Dial 8 Port)
Digital telephone sets	Yes	No
FCAPS	Yes	Yes, extended to include SRG-specific alarms and keycodes
Network Configuration Manager	Yes	Yes
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) (CS 1000 Release 4.0 requires patch MPLR22418)
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	N/A	Yes

Supported devices

The SRG50 Release 5.0 supports:

- IP Phones 1110, 1120E and 1140E
- IP Phone 2001, 2002, 2004, and 2007
- IP Audio Conference Phone 2033
- IP Softphone 2050

- Mobile Voice Client (MVC) 2050
- WLAN Handsets 2210, 2211, 2212, 6120, and 6140
- IP Phone 1210, 1220, and 1230 (CS1000 Release 5.5 and above)
- Analog (500/2500 type) telephones
- Analog devices such as fax machines

Note: Throughout this document, the IP Phones in this list are referred to collectively as IP Phones.

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 table SRG50 terminology on page 18 identifies SRG terms that may be unfamiliar to main office installers. They are provided to facilitate communications between SRG and main office personnel. In the table, the Business Element Manager path where the term appears is provided for reference; the paths are provided for reference and may not represent every appearance of the term.

Table 5 SRG50 terminology (Sheet 1 of 2)

Term	Description
Port	For telephony configuration (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 (Configuration > Resources) and data services (Configuration > Data Services), port is used in the context of the TCP/IP protocol suite.
IP Terminal	IP telephone Configuration > Resources > Telephony Resources > IP & App 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 Configuration > Resources > Telephony Resources > IP & App Sets Mapping DN to Telephone Configuration > Telephony > Sets DN 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 SRG and provide incoming and outgoing communication paths. Paths can be physical (examples: loop; PRI; T1) or virtual (VoIP trunks). Configuration > Resources

Table 5 SRG50 terminology (Sheet 2 of 2)

Term	Description
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. Configuration > Telephony > Lines
Physical Lines	Physical trunks. Configuration > Telephony > Lines > Active Physical Lines (Lines 061 to 124)
VoIP Lines	VoIP trunks. 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. Business Element Manager provides two methods for assigning target lines to DNs. 1) Configuration > Telephony > Sets > All DNs > All DNs table > Details for DN subpanel > Line Assignment tab or 2) 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).
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 SRG branch office requires datafill at both the SRG and the main office call server. Main office configuration drives SRG configuration, and Nortel recommends that the main office activities be concluded before undertaking SRG configuration.

For information specific to Nortel Communication Server 1000, see CS 1000 considerations on page 31. For more information about configuring the SRG 50 5.0 for a main office, see *Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50*.

SRG operating modes

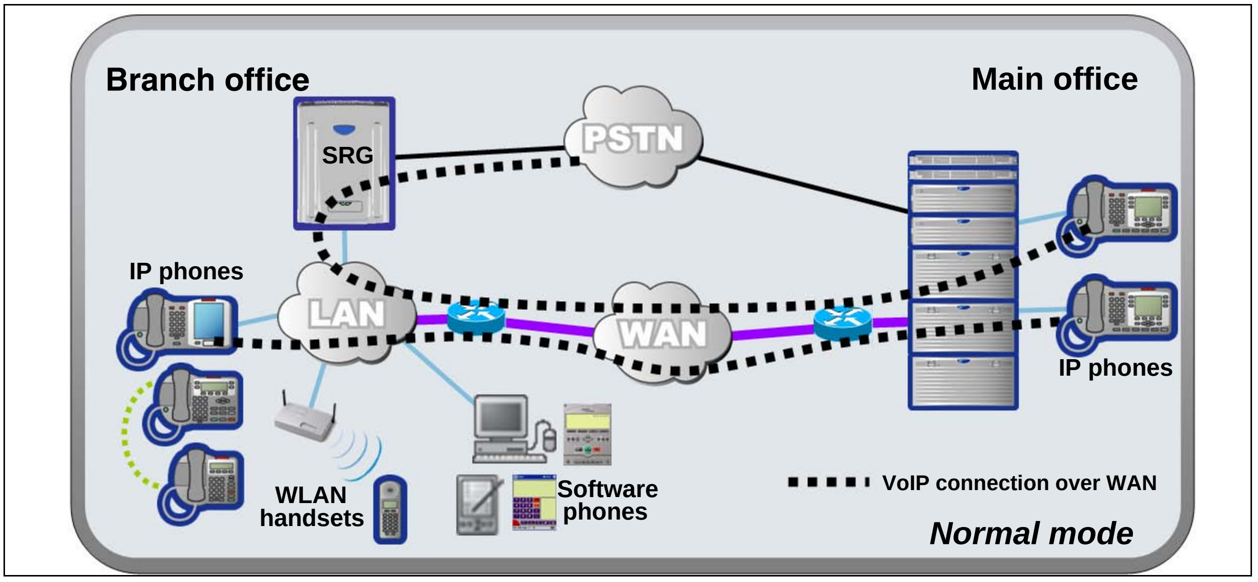
The SRG has two operating modes:

- Normal mode
- Local mode

Normal mode

In normal mode (see the figure Normal mode on page 20), the SRG is connected to the main office call server over a WAN using VoIP trunks. From the perspective of the main office, the SRG is a branch office.

Figure 1 Normal mode



IP telephones connected at the SRG 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 SRG PSTN. For main office callers, the SRG 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 (either H.323 or SIP) to the main office, which terminates the call at the branch user. Similarly, calls from analog telephones connected to the SRG 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 SRG 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 SRG. This means that these calls do not use any WAN bandwidth between the main office and the branch office after calls are established.

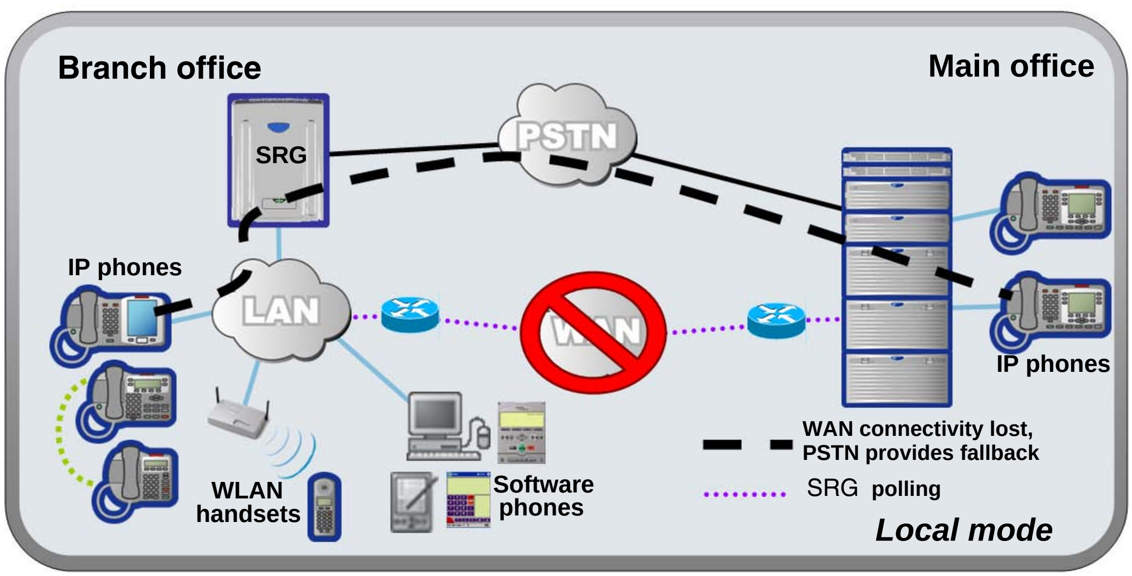
When a branch user IP telephone calls a main office IP telephone and vice versa, the call is a simple station-to-station call within the main office call server. Since the branch user IP telephone is physically remote from the call server, the media path goes through the WAN connection between the main office and the SRG, 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 IP Phones connected to the SRG revert to local mode automatically. In local mode, the IP users connected to the SRG are under the control of the SRG. When in local mode, main office call features are not available to users attached to the SRG. The SRG 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, see Features in local mode on page 60. Local mode is illustrated in the figure Local mode on page 21.

Note: The IP Phone KEM is supported on an SRG with normal mode IP Phones. It does not function with local mode or test local mode IP Phones.

Figure 2 Local mode



The SRG handles all call processing. Calls between two IP telephones at the SRG are handled locally as a simple station-to-station call. When an IP telephone initiates a local PSTN call, the SRG routes the call to a trunk that is connected to the local PSTN. Incoming DID calls are also handled by the SRG 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 SRG, and is in local mode. When the SRG configuration at the main office and the SRG 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.

Failure to register with the main office

When configured as a branch office user set, an IP telephone at the SRG 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 SRG is redirecting it back to the main office because connectivity has been reestablished (see Loss of WAN or VoIP connectivity on page 22).
- 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 87.

Loss of WAN or VoIP connectivity

The WAN or VoIP connectivity between the main office and the SRG 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 connection with the main office terminal proxy server or Centrix IP client manager. The IP telephones reboot and reregister at the SRG, 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 SRG 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 SRG 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 SRG 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 (see CS 1000 considerations on page 31).

On-Site Notification

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

The OSN is enabled by a third-party tool that connects to the SRG through a LAN CTE interface. A single LAN CTE interface must be enabled on the SRG by applying an keycode. (The LAN CTE interface is enabled by default by the SRG 50 Release 5.0 Authorization keycode.) For more information, see LAN CTE Configuration Guide (NN40010-601).

SRG installation and configuration summary

The *SRG50 5.0 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 *SRG50 5.0 Configuration Guide* where applicable.

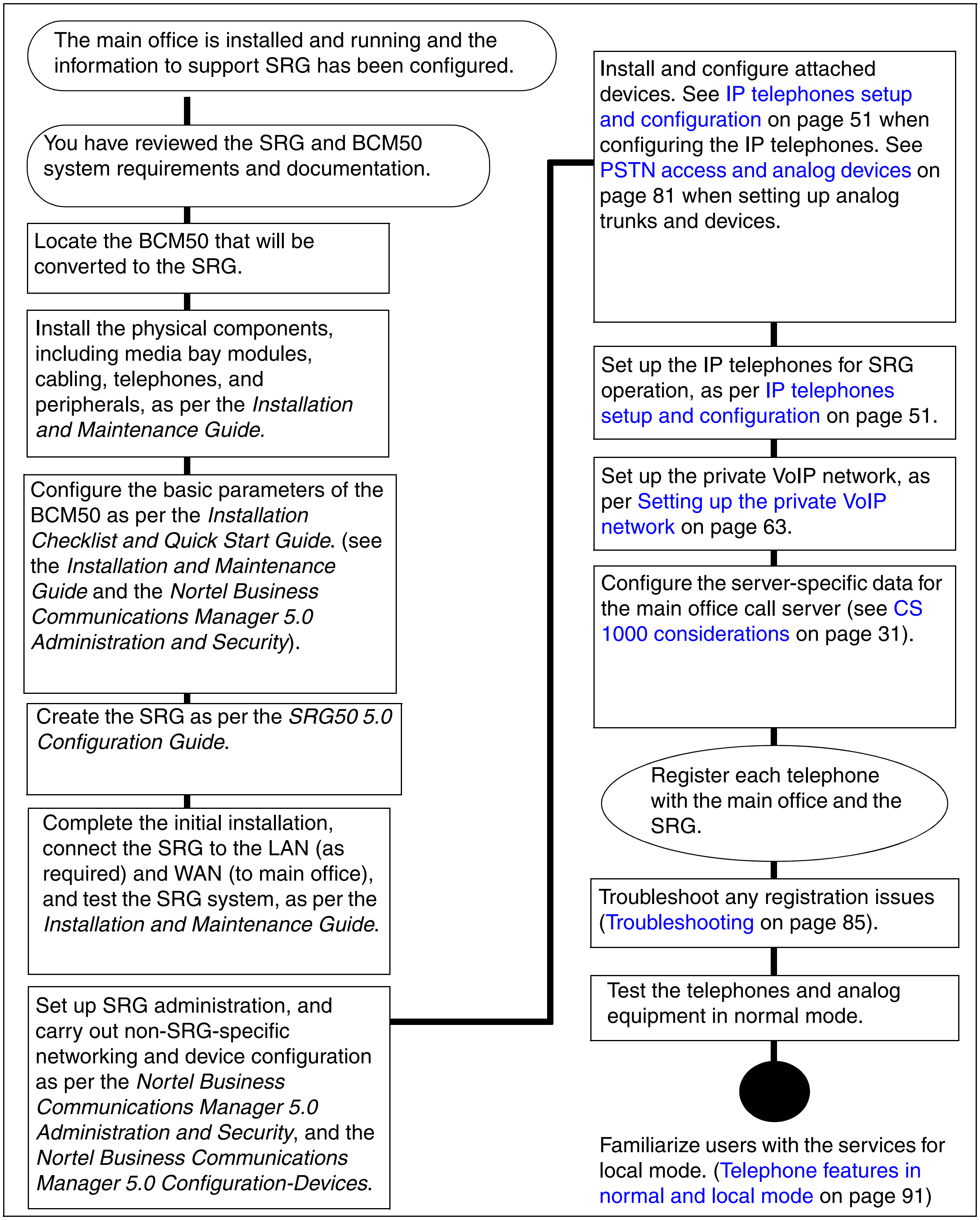
Generally, SRG50 activities leverage an installer's general knowledge of BCM50 activities. However, Nortel recommends that the BCM50/SRG50 site engineer and installer familiarize themselves with SRG-specific requirements before starting any installation activities.

The figure Process map for installing and configuring an SRG on page 24 provides a process map for installing and configuring an SRG50. The procedures in the *SRG50 5.0 Configuration Guide* assume that the following activities have been completed:

- The BCM50, including 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.
- CS1000 main office system has been installed and configured to support SRG.
- The SRG 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 (for information about configuring IP Phones, see IP telephones setup and configuration on page 51).
- Non-SRG-specific networking and device configuration has been completed (for information about configuring the network, see Setting up the private VoIP network on page 63).

Process map for installing and configuring an SRG

Figure 3 Process map for installing and configuring an SRG



Chapter 3
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 Business Element Manager panels.

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.

1 Configure the SRG IP address, net mask, and gateway.

Configuration >System > IP Subsystem

External Reference: *Installation and Maintenance Guide*

2 Confirm the number of IP sets and VoIP trunks.

Configuration > Resources > Application Resources

The **Licence** column indicates the number of resources available.

External Reference: *Keycode Installation Guide*

3 Verify the global telephony settings.

Configuration > Telephony > Global Settings

External Reference: *Nortel Business Communications Manager 5.0 Configuration—Devices*

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

Administration > Utilities > Reset > Cold Reset Telephony Services button > Cold Reset Telephony dialog box > Start DN field

Internal Reference: Basic parameters on page 65

External Reference: *Installation and Maintenance Guide*

5 Verify the DN length.

i) For local calls between telephones on the SRG.

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 subpanel > 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. See *Nortel Business Communications Manager 5.0 Configuration—Telephony*.)

Internal Reference: Basic parameters on page 65

External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

6 Verify the line pool assignment of VoIP trunks.
In the default configuration, the VoIP trunks are assigned to line pool BlocA. Instructions in the *SRG50 5.0 Configuration Guide* assume that the default configuration has been maintained.

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

External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

7 The SRG 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 *SRG50 5.0 Configuration Guide* assume that the default configuration has been maintained.

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

External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

* Category 1 countries

IP telephone configuration

1 Configure the registration password.

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

Internal Reference: Registration password on page 51

2 Configure the local mode indication (Advertisement/Logo).

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

Internal Reference: Local mode indication on page 53

3 Configure the IP telephone codec and jitter settings.

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

Internal Reference: IP telephone codec and jitter settings on page 53

4 Configure the telephone (DN) records.

Configuration > Telephony > Sets > All DNs

Internal References: Telephone (DN) records configuration on page 54

5 Configure the received numbers.

Configuration > Telephony > Lines > Target Lines

Internal Reference: Received numbers configuration on page 57
External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*.

6 Decide on the call forwarding option.

Internal Reference: Call forwarding options on page 58

7 Configure the IP telephones.

Internal Reference: Configuration settings for redirected phones on page 59

Dialing plan configuration

1 Configure the private network type (CDP or UDP).

Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > Private Network Settings subpanel > Private network type

Internal Reference: Private dialing plan on page 66
External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

2 Enable MCDN TAT.

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

Internal Reference: Meridian Customer Defined Network (MCDN) on page 67
External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

VoIP trunk configuration

1. Configure VoIP trunk QoS settings.

   Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Media Parameters tab or SIP Media Parameters tab

   Internal Reference: QoS settings (codec, jitter buffer, and related items) on page 68
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

2. Enable or disable fallback.

   Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Settings tab or SIP Settings tab

   Internal References:
   Fallback configuration on page 70
   SRG PSTN access on page 75
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

3. Configure gatekeeper settings.

   Configuration > Resources > Telephony Resources > Modules panel > IP Trunks row > H323 Settings tab

   Internal Reference: Gatekeeper routing on page 71
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

4. 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 72
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

5. 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 72
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

6. Assign remote access packages to the VoIP trunks.

   Configuration > Telephony > Call Security > Remote Access Packages

   Internal Reference: SRG PSTN access on page 75
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

Call routing configuration

1. Decide on the fallback scheme.

   Internal Reference: Fallback configuration on page 70

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

   Configuration > Telephony > Dialing Plan > Routing

   Internal Reference: Outgoing calls configuration on page 73
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

3. Configure access to the SRG PSTN (for both local and tandem calls).

   Configuration > Telephony > Dialing Plan > Routing

   Internal Reference: SRG PSTN access on page 75
   External Reference: *Nortel Business Communications Manager 5.0 Configuration—Telephony*

4. Configure for Network Bandwidth Management and Advanced Network Bandwidth Management.

   Internal Reference: Bandwidth management configuration: NBWM, ADBWM, and ACR on page 38
   External Reference: *Branch Office: Installation and Configuration* (553-3001-214) and *Data Networking for Voice over IP* (553-3001-160)

5. Configure for Alternative Call Routing.

   Internal Reference: Bandwidth management configuration: NBWM, ADBWM, and ACR on page 38
   External Reference: *What's New for Communication Server 1000 Release 4.5* (553-3001-015)

Redirection and call forward configuration

1. Configure the main office settings.

   Configuration > Resources > Survivable Remote Gateway > S1000 Main Office Settings tab

   Internal Reference: CS 1000 information for the SRG on page 42

2. Configure the IP terminal settings.

   Configuration > Resources > Survivable Remote Gateway > S1000 IP Terminal Details tab

   Internal Reference: IP telephones redirection on page 45

Chapter 4
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). SRG50 Release 5.0 operates with CS 1000 running Release 5.0 and later.

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.

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

- "CS 1000 and SRG terminology comparison" on page 31
- "Normal and local mode overview" on page 33
- "Virtual trunk capacity" on page 36
- "Vacant Number Routing (VNR)" on page 36
- "Bandwidth management" on page 37
- "Bandwidth management configuration: NBWM, ADBWM, and ACR" on page 38
- "Emergency Services Access (ESA) configuration" on page 41
- "CS 1000 information for the SRG" on page 42
- "IP telephones redirection" on page 45
- "Firmware upgrade" on page 49

CS 1000 and SRG terminology comparison

The table Comparison of CS 1000 and SRG terms and contexts on page 31 compares configuration-related terms and contexts of the CS 1000 and the SRG.

Table 6 Comparison of CS 1000 and SRG terms and contexts (Sheet 1 of 3)

Term or Context	CS 1000	SRG
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 SRG in the MOTN field (see "IP telephones redirection" on page 45).	

Table 6 Comparison of CS 1000 and SRG terms and contexts (Sheet 2 of 3)

Term or Context	**CS 1000**	**SRG**
	BUID (branch user ID) The dialable number of an IP telephone at the SRG when it is called from a phone located at the main office or another branch office.	The CS 1000 BUID is entered on the SRG (see S1000 IP Terminal Details panel on page 47) but there is no SRG 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 SRG when it is called from another phone at the SRG.
	In the case of a CDP dialing plan, it is recommended that the BUID and the SRG 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 SRG DN be the same as <DN>. The dialable number of an IP telephone at the SRG, when dialed from another phone at the SRG, remains the same in both normal and local mode if the preceding recommendations are implemented.	
	AC1	VOIP Trunk Access Code (see CS 1000 information for the SRG on page 42) Destination code for VoIP trunks (see Outgoing calls configuration on page 73)
	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
Access codes (SRG: 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 = SRG Zone ID ZDP = VPNI. That is, the CS 1000 Zone ID is entered on the SRG in the Zone ID field, and the ZDP is entered on the SRG in the VPNI field (see Bandwidth management on page 37).	
Trunks	public exchange	PSTN

Table 6 Comparison of CS 1000 and SRG terms and contexts (Sheet 3 of 3)

Term or Context	CS 1000	SRG
IP telephone password	installer password	global password
	The two passwords can be made the same. See Registration password on page 51.	

Normal and local mode overview

Normal mode and local mode overview provides a description of the following sections:

- Normal mode
- Local mode
- Survivability
- Recovery to normal mode
- Local mode operation
- Virtual trunks

Normal mode

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

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

Local mode

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

1 IP Phone 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.

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

In the event that the IP Phones at the branch office lose the connection to the main office CS1000 call server for any reason (WAN failure, main office call server failure, main office Signaling Server failure), the SRG50 reverts to local mode automatically. Essentially, when VoIP connectivity is lost, each IP Phone loses its Reliable UDP (RUDP) connection with the main office terminal proxy server (TPS). The IP Phones at the branch office reboot and re-register to the SRG50, placing them in local mode.

Once this has occurred, the IP Phones displays an indication on the display area that the set is in local mode of operation. This display is configurable by installers to meet local language and usage norms.

In local mode, the IP users connected at the branch office are under the control of the SRG50 call services. As such, the normal main office call server features are not available. The SRG50 offers a basic feature set when in local mode which allows IP Phones to continue to make and receive calls internally within the branch office and over the provisioned local PSTN interfaces. Basic services, such as transfer, last number redial, and single key access through the PSTN to a centralized voice messaging system are supported. Local PSTN access and local Emergency Services access is also supported. No local applications or Business Communication Manager features are supported in local mode operation.

Analog devices continue to be under the control of the SRG50 system. It is the intent of local mode to provide continued access to the PSTN for critical calls and emergency services.

In local mode, since the SRG50 handles all call processing, calls between two IP phones at the SRG50 are handled locally as a simple station-to-station call. When an IP Phone 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 Phone.

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

While in local mode, the SRG50 system continues to monitor for a main office CS1000 heartbeat signal, and once detected, automatically redirects phones on an individual basis back to normal mode of operation. If a call is active, the SRG waits until the call is completed before redirecting the phones; calls in progress are not interrupted. This switch-hitter occurs almost immediately once the SRG determines that an individual phone can be redirected. This reinstates the CS1000 normal user interface and feature set for the IP Phone user, on a user by user basis.

The SRG50 system implements the same interface used by the MG1000B system to interact with the main office CS1000 system. This allows the main office to identify attached clients and the local PSTN as branch office entities, enabling proper operation of dial plans and E911 access.

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

Survivability

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

SRG supports the Geographic Redundancy feature.

In the event of a WAN failure, the SRG IP Phones lose communication with the main office. This causes the SRG IP Phones to reset and register with the SRG. The IP Phones then operate in local mode, providing basic telephony services delivered by the local SRG system.

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, the IP Phone attempts to communicate with the SRG. After communication with the SRG is established, the SRG redirects the IP Phone to the main office. When the SRG IP Phone attempts to register with the main office, the SRG first queries the Primary 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 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 Virtual Trunks are not available. (SRG analog [500/2500-type] telephones are accessible if alternate routing is available through the PSTN.)

For more information, see *Nortel Business Communications Manager 5.0 Configuration—Telephony* and *Nortel Business Communications Manager 5.0 Planning and Engineering*.

Recovery to normal mode

Once communication is re-established with the main office call server, all IP Phones at the branch office that are in local mode automatically redirect and reregister to the main office and return to normal mode operation. IP Phones that were busy at the time communication was reestablished complete the call in local mode, and then reregister with the main office after the call is complete.

Testing the telephone in local mode

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

Nortel recommends testing local mode operation after changing the provisioning for a telephone on the SRG.

To ensure that users do not forget to resume normal mode operation, the SRG redirects the telephone to the main office to return the telephone to normal mode. This occurs if the telephone remains registered to the SRG in test local mode for ten minutes (default setting). Alternatively, the user can press the Quit key from the set to return to normal mode.

Virtual trunk capacity

The SRG supports 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 32.

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 32.

Vacant Number Routing (VNR)

The SRG does not support Vacant Number Routing (VNR). Instead, the SRG 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 SRG to route all calls not terminated locally by the SRG. 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 feature is similar to the VNR feature on the Media Gateway 1000B (MG1000B).

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 SRG must dial a separate destination code before the main office DN.

For details about dialing plan and routing configuration, see Setting up the private VoIP network on page 63.

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 SRG interoperates with the Network Bandwidth Management (NBWM) feature in a manner similar to Media Gateway (MG) 1000B, though only G.711 and G.729 codecs are supported. At the SRG, a Virtual Private Network ID (VPNI) and Zone ID are entered with values defined by the main office configuration (see Bandwidth management configuration: NBWM, ADBWM, and ACR on page 38). The VPNI and Zone ID allow the CS 1000 to recognize that H.323 calls to and from the SRG 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 SRG in order to implement he Adaptive Network Bandwidth Management (ADBWM) feature on the SRG (see Bandwidth management configuration: NBWM, ADBWM, and ACR on page 38).

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 SRG 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 (see Bandwidth management configuration: NBWM, ADBWM, and ACR on page 38).

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.

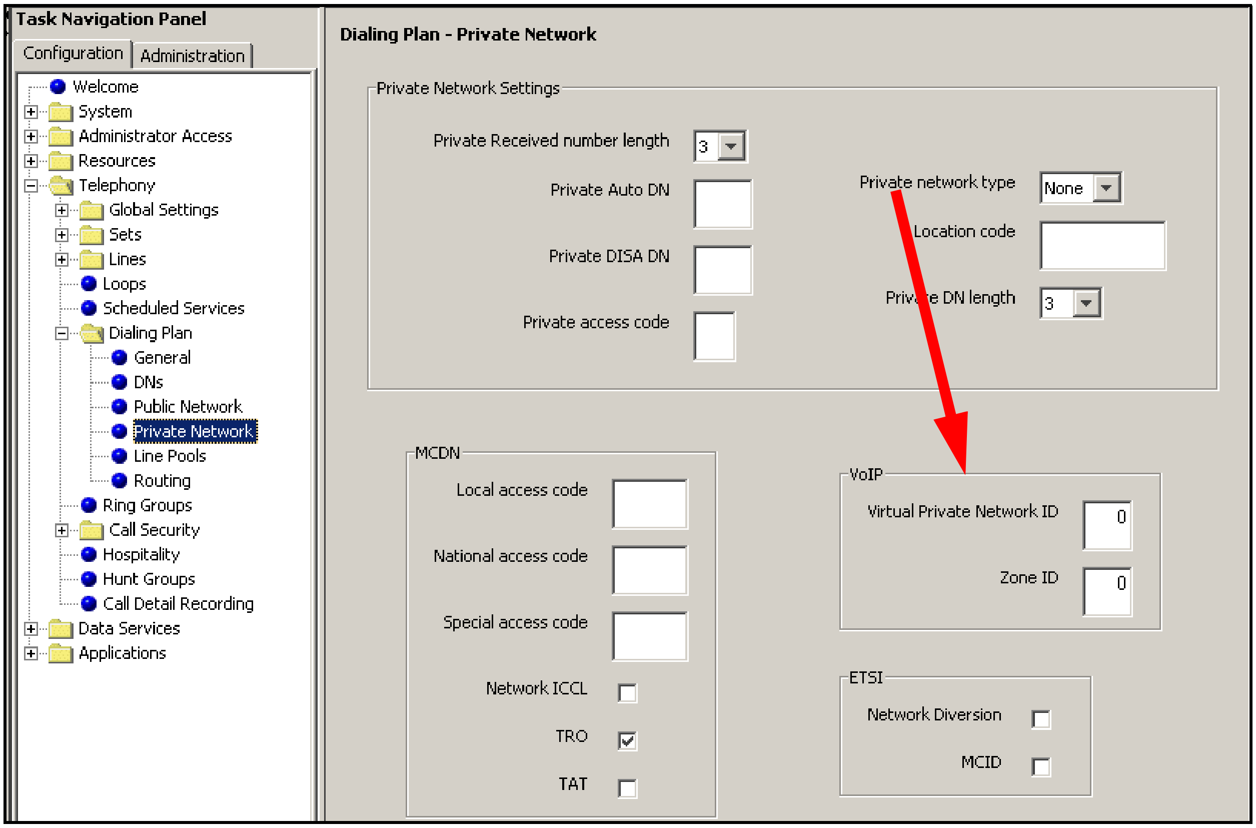
Network administrators who do not want calls to be blocked, yet have a limited amount of bandwidth available, can use ACR to overflow calls to conventional trunks (PSTN or Tie/MCDN). ACR allows calls to be routed by overflowing them, trading off the capital cost of WAN bandwidth against the incremental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

Bandwidth management configuration: NBWM, ADBWM, and ACR

To configure SRG for NBWM and ADBWM

1. Obtain the Virtual Private Network ID and the Zone ID numbers configured at the main office.
2. Use Business Element Manager to enter these numbers in the appropriate fields at Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > VoIP subpanel (see the figure Dialing Plan - Private Network panel, VoIP subpanel on page 38).

Figure 4 Dialing Plan - Private Network panel, VoIP subpanel



To configure Alternative Call Routing with attendant service

1 Complete the procedure, To configure SRG for NBWM and ADBWM on page 38.

For more information, see Outgoing calls configuration on page 73, and see also *Nortel Business Communications Manager 5.0 Configuration—Telephony*.

2 Obtain the ALTPrefix for the SRG (configured at the main office).

3 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** list; default is **A**).

4 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 In 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 list.

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 with DID trunks

Note: In order to support DID trunks, the MODN dialed by an SRG 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 SRG for NBWM and ADBWM on page 38.

For more information, see Outgoing calls configuration on page 73 and the *Nortel Business Communications Manager 5.0 Configuration—Telephony.*

2 Obtain the ALTPrefix for the SRG (configured at the main office).

3 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** list; default is **A**).

4 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 list.

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.

Emergency Services Access (ESA) configuration

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

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

For IP telephones in local mode, and for other telephones at the SRG, see the *Networking Configuration Guide* for configuring emergency services.

To configure Emergency Services Access

1 Verify that a remote access package has been assigned to the VoIP trunks (see Remote Access Package for VoIP trunks on page 75).

2 Obtain the ESA Special Number (SPN).

3 In Business 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 SRG 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 SRG 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 and then change the value to 3.

CS 1000 information for the SRG

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

S1000 Main Office Settings panel

The table S1000 Main Office Settings on page 42 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 7 S1000 Main Office 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 7 S1000 Main Office Settings (Sheet 2 of 3)

Field	Values	Description
Network Connect Server Port	16500 (default) Range: 0 to 65535	Port on the SRG used to connect to the NCS.
Heartbeat Protocol Port	16501 (default) Range: 0 to 65535	Port on the SRG that the SRG 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. see "Outgoing calls configuration" on page 73. ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>
Test Local Mode Timeout	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. see the *Networking Configuration Guide* for naming conventions.	Gatekeeper setting that identifies the SRG. 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 (see VoIP trunking configuration on page 69). This applies both to H323 or SIP trunks.

Table 7 S1000 Main Office Settings (Sheet 3 of 3)

Field	Values	Description
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.
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 SRG 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.

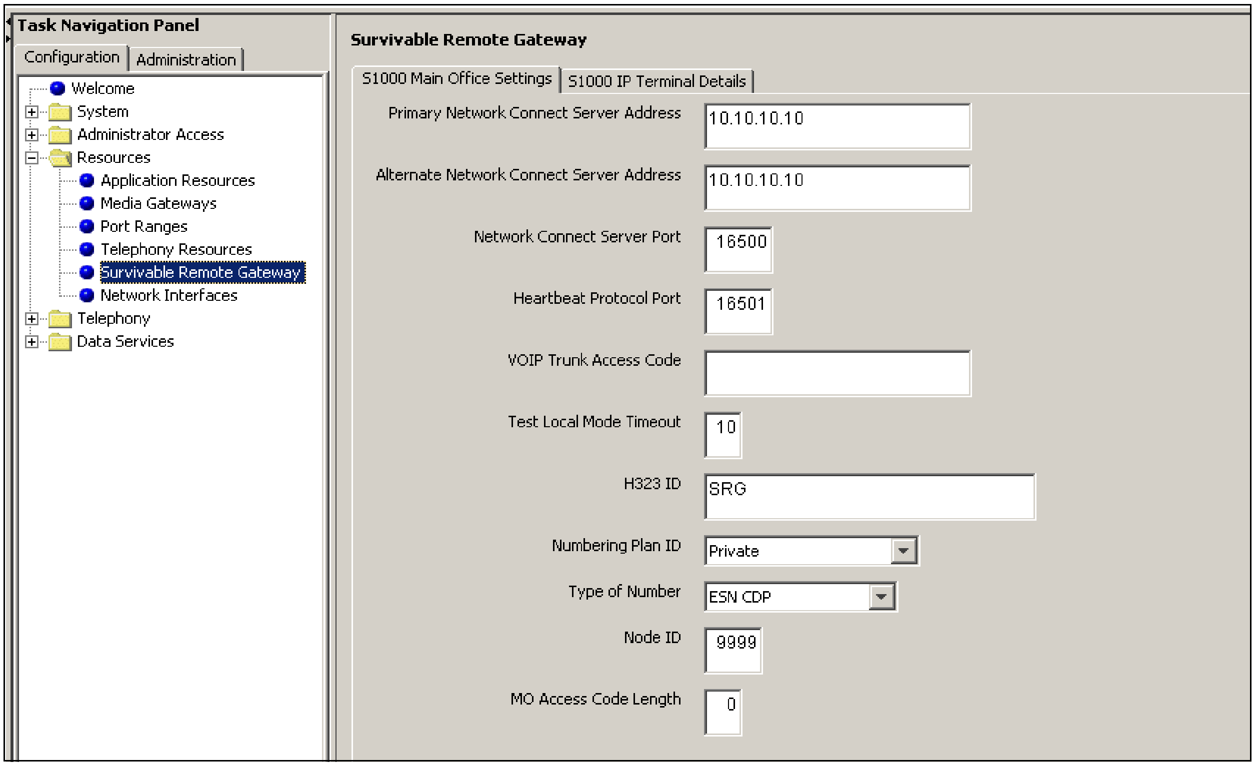
To datafill the S1000 Main Office Settings panel

1 In Business Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway** (see the figure S1000 Main Office Settings panel on page 45).

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

3 Enter the information in the appropriate fields.

Figure 5 S1000 Main Office Settings panel



IP telephones redirection

Once an IP telephone at the SRG is configured (see IP telephones setup and configuration on page 51), it automatically registers with the SRG (S1). To configure an IP telephone for redirection to the main office call server, SRG-specific datafill is required. The SRG-specific configuration includes:

- IP telephones numbers and models on page 45
- S1000 IP Terminal Details panel on page 47

IP telephones numbers and models

SRG DNs are assigned to IP telephones using Business Element Manager. Redirection to the main office requires configuration at the SRG to associate the SRG 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 SRG, the actual IP telephone configured to an SRG DN must be the same model that is configured in the TN record that is associated with the SRG DN.

The table IP telephone numbers and models on page 46 provides a record of the CS 1000 TN and BUID, the SRG MOTN and DN, and the model of phone, to facilitate installation and provide a permanent record of the configuration.

Table 8 IP telephone numbers and models

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

S1000 IP Terminal Details panel

Business Element Manager provides SRG-specific panels for recording the CS 1000 TN and BUID that are associated with a particular SRG DN. The table SRG S1000 IP Terminal Details fields on page 47 lists and describes the fields on the **S1000 IP Terminal Details** panel.

Table 9 SRG S1000 IP Terminal Details fields

Field	Values	Description
DN	Read-only	The SRG DN assigned to the telephone. The SRG DN must be configured before proceeding with the procedures that follow in this section. See Telephone (DN) records configuration on page 54.
Hardware ID	Read-only	Hardware ID. Unique for each IP telephone.
Status	Read-only	Current status of the telephone. See IP terminal details on page 86 (expand the field to read the entire status message).
Firmware Version	Read-only	Updated by the main office when a terminal is sent back to the SRG 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 SRG when it is called from a phone located at the main office or another branch office. The BUID at the SRG 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 SRG DNs must be configured before the following procedures can be undertaken. See Telephone (DN) records configuration on page 54.

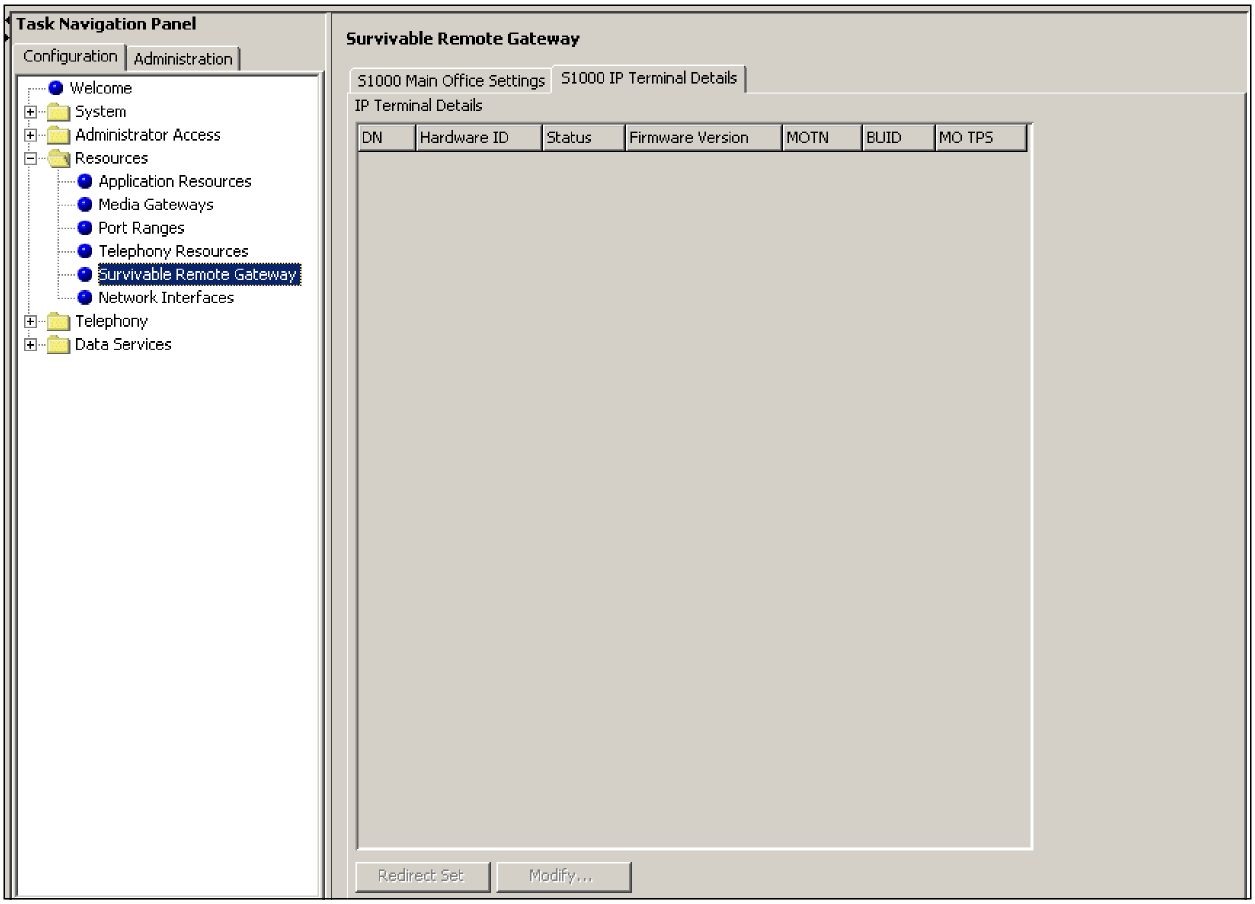
To enter the MOTN and BUID

1. In Business Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.
2. Select the **S1000 IP Terminal Details** tab (see the figure S1000 IP Terminal Details panel on page 48).
3. Refer to the numbers and models recorded in the table IP telephone numbers and models on page 46.
4. Select the required DN.
5. Press the **Modify** button.
6. Enter the MOTN and the BUID in the appropriate fields.

To redirect the telephone to the main office call server

1. In Business Element Manager, navigate to **Configuration > Resources > Survivable Remote Gateway**.
2. Select the **S1000 IP Terminal Details** tab (see the figure S1000 IP Terminal Details panel on page 48).
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 table Configuration settings for redirected IP Phones on page 49. The *Nortel Business Communications Manager 5.0 Configuration—Devices* and the *Nortel Business Communications Manager 5.0 Configuration—Telephony* provide detailed instructions for configuring IP telephones.

Table 10 Configuration settings for redirected IP Phones

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

Firmware upgrade

The redirected IP telephones at the SRG 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.

Supported firmware

The table Supported IP clients and firmware versions on page 49 lists the IP clients and related firmware versions supported on the SRG50. The SRG50 column indicates the firmware versions included with the SRG software. The CS 1000 columns identify the version of firmware to use for specific releases.

Table 11 Supported IP clients and firmware versions (Sheet 1 of 2)

IP Client	SRG50	CS 1000		
		Release 5.0	**Release 5.5 (apply via Enhanced Firmware Download feature)**	**Release 6.0 (apply via Enhanced Firmware Download feature)**
Phase I: 2002, 2004*	0602B76 or 0603B76	0602B76 or 0603B76	0602B76 or 0603B76	0602B76 or 0603B76
Phase II: 2001, 2002, 2004	0604DCL or greater	0604DCL or greater	0604DCL or greater	0604DCL or greater

Table 11 Supported IP clients and firmware versions (Sheet 2 of 2)

IP Client	SRG50	CS 1000		
		Release 5.0	**Release 5.5 (apply via Enhanced Firmware Download feature)**	**Release 6.0 (apply via Enhanced Firmware Download feature)**
IP Phone 2007	C6M or greater	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
WLAN Handsets 2210/2211	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software
WLAN Handsets 6120/6140	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software
WLAN Handset 2212	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software	Not embedded in SRG software
IP Audio Conference Phone 2033	S12 or greater	S12 or greater	S12 or greater	S12 or greater
IP Phone 1110	0623C60	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
IP Phone 1120E	0624C60	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
IP Phone 1140E	0625C60	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
IP Phone 1200 series	062AC60	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
IP Softphone 2050	build 385 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
MVC 2050	build 126 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater	Unistim 3.3 or greater
*These phones may work with SRG50 5.0, however, they are End of Life and are no longer supported.				

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 SRG for upgrading. If the required file does not exist on the SRG, or its version is incorrect, the SRG initiates an SFTP (if using CS1000 Release 6.0) or FTP session to the TPS for that phone to retrieve the required file. The SRG upgrades the phone and redirects it back to the CS 1000.

Chapter 5
IP telephones setup and configuration

IP telephone setup and DN configuration are described in detail in the *IP Telephone Installation and Configuration Guide* and *Nortel Business Communications Manager 5.0 Configuration—Telephony*, respectively. SRG-specific procedures and settings include:

Registration password

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

To set the IP telephone registration password

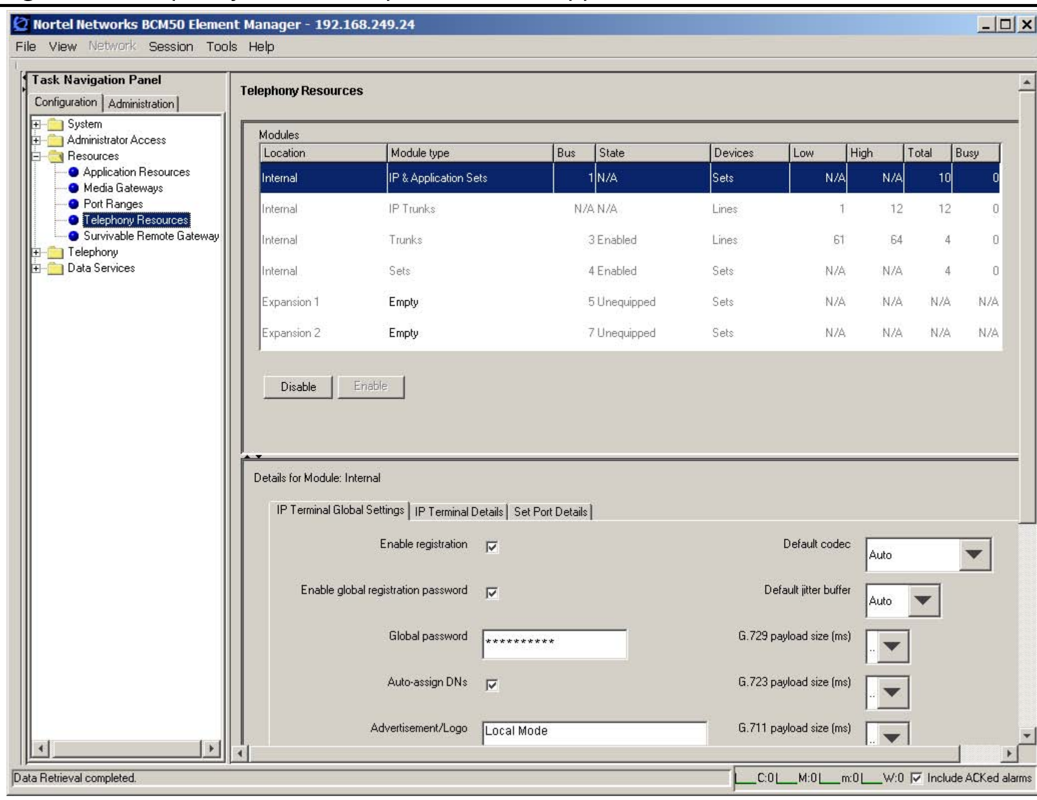
1. In Business Element Manager, navigate to **Configuration > Resources > Telephony Resources**.
2. On the **Modules** panel, locate the **Module type** column and select the **IP Sets** row (see the figure Telephony Resources panel, IP and App Set on page 52).
3. On the **Details for Module:** subpanel, select the **IP Terminal Global Settings** tab.

There are three fields that define the password registration process (see the table Password registration parameters on page 52):
Enable registration, **Enable global registration password**, and **Global password**.

Table 12 Password registration parameters

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:** Nortel recommends that you synchronize this password with the CS 1000 password.

Figure 7 Telephony Resources panel, IP and App Set



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 SRG automatically assigns DNs, see the *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 In Business Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

2 On the **Modules** panel, locate the **Module type** column and select the **IP Sets** row (see the figure Telephony Resources panel, IP and App Set on page 52).

3 On the **Details for Module** 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 SRG environment. They do not have to be the same as the main office settings (in contrast to the QoS settings for the VoIP trunks; see QoS settings (codec, jitter buffer, and related items) on page 68).

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

1 In Business Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

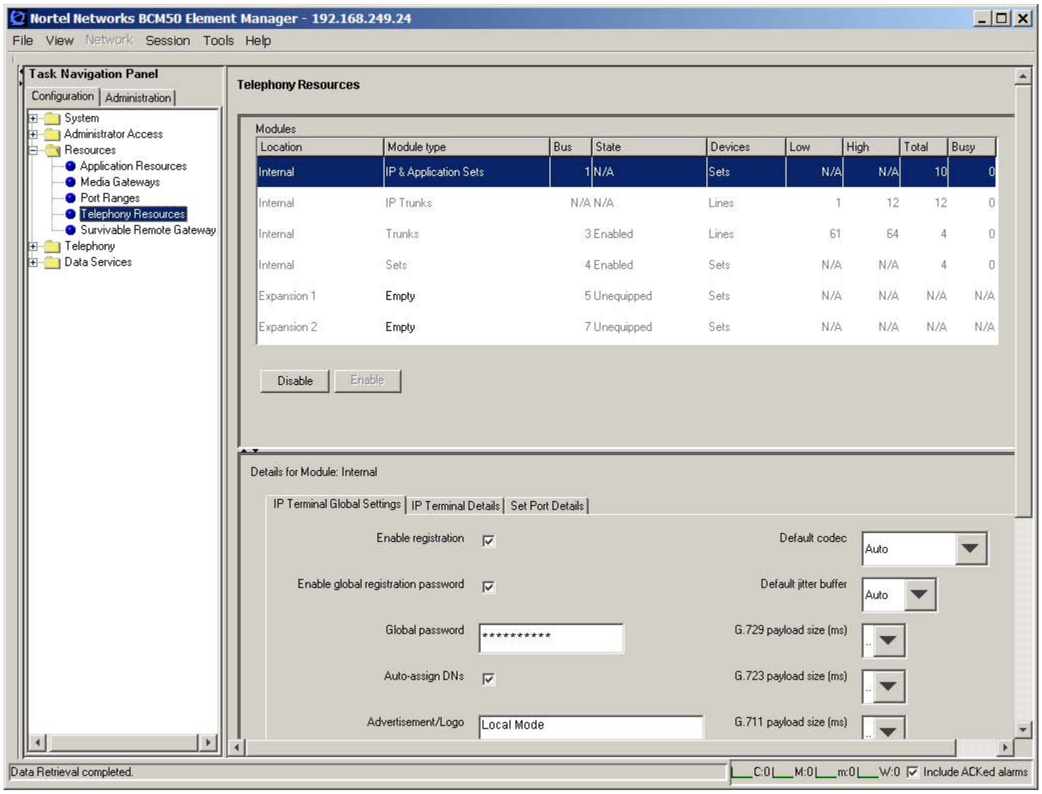
2 On the **Modules** panel, locate the **Module type** column and select the **IP Sets** row (see the figure Modules panel, IP & Application Sets on page 54).

3 On the **Details for Module** 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 8 Modules panel, IP & Application Sets



Telephone (DN) records configuration

DN records for IP telephones are configured through the **All DNs** panel (**Configuration > Telephony > Sets > All DNs** (see the figure All DNs panel on page 56). The *Nortel Business Communications Manager 5.0 Configuration—Telephony* and the *Nortel Business Communications Manager 5.0 Configuration—Devices* 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. For more information, see SRG50 overview on page 15.

To configure DN records for redirected IP telephones

1. In Business 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 (see CS 1000 considerations on page 31).

5 From the **Model** list, select the model of telephone assigned to this DN.

For CS 1000: See IP telephones numbers and models on page 45.

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 SRG can forward all calls to the main office call server automatically (see Call forwarding options on page 58). In local mode, the Call Forward All Calls feature is automatically discontinued and the SRG routes calls to the SRG 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 (see the figure All DNs panel on page 56).

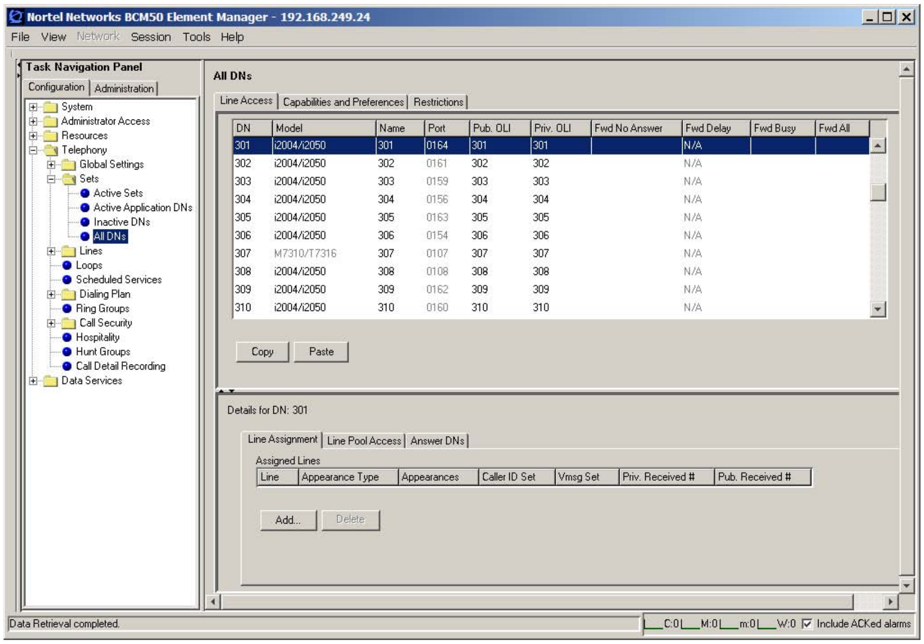
10 Ensure that the **Appearance Type** (**Details for DN** details panel > **Line Assignment** tab) is set to **Ring only**.

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). For a description of target lines, see SRG50 terminology on page 18.

Note: At this point, you may want to configure the received numbers. Refer to Received numbers configuration on page 57 and then return to this procedure.

Figure 9 All DNs panel



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

14 In the **Intercom keys** column, 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
1120E	08
1140E	08

Received numbers configuration

The **Public Received number length** and the **Private Received number length** (see Basic parameters on page 65) determine the number of digits that the SRG 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, see SRG50 terminology on page 18).

> **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. In Business Element Manager, navigate to **Configuration > Telephony > Lines > Target Lines.**
2. Select the target line of the DN for which you want to configure the received numbers.
3. Double-click the **Pub. Received #** field, and then enter the retained received digits for calls originating from the PSTN.
4. Double-click the **Priv. Received #** field, and then 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.

DHCP settings configuration

To configure DHCP settings for SRG operation

1. In Business Element Manager, navigate to **Configuration > Data Services > DHCP Server**.
2. Select the **General Settings** tab.
3. From the **DHCP Server is** list, select **Enabled - IP Phones Only**.
4. From the **WINS node type** list, 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 tab 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 SRG.
   b. From the **Port** list, select **SRG**.
   c. In the **Retry count** field, enter the number of retries that the IP telephone is allowed to connect to the SRG before an event is generated (see IP terminal details on page 86).

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 SRG:

- The target DN is determined by the BUID.

  In this case, the information required for call forwarding is entered using the SRG-specific panels of Business Element Manager (**Configuration > Resources > Survivable Remote Gateway**). See CS 1000 considerations on page 31.

> **Note:** Call forwarding is mandatory for CS 1000.

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

  The DN is configured through Business Element Manager on the **Telephony > Sets > Active Sets** panel using the standard BCM50 procedure (for details, see the *Nortel Business Communications Manager 5.0 Configuration—Devices*). Typically, the call forward number is the BUID.

  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 SRG-specific panels (the SRG-specific panels **must** be completed). The SRG looks at the SRG-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 *Nortel Business Communications Manager 5.0 Installation—Devices* and the *Nortel Business Communications Manager 5.0 Configuration—Telephony* 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 table Configuration settings for redirected IP telephones on page 59.

Table 13 Configuration settings for redirected IP telephones

Parameter	CS 1000
S1 IP	SRG IP address
S1 Port	7300
S1 Action	1
S1 Retry Count	1
S2 IP	SRG IP address
S2 Port	7300
S2 Action	1
S2 Retry Count	1

For more information, see CS 1000 considerations on page 31.

Test Local Mode

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

For more details on invoking test local mode, see S1000 Main Office Settings panel on page 42.

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 the WLAN Handsets 2210/2211, pressing the End key causes the phone to exit Test Local Mode.

Features in local mode

In local mode, IP telephones at the SRG no longer have access to the full suite of main office applications. However, the SRG does provide a set of features that include connectivity with the local PSTN, access to Emergency Services, and the ability to call local extensions. For more information about call features available in local mode, refer to the *Nortel Communication Server 1000 Main Office configuration for Survivable Remote Gateway 50.*

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: **Configuration > Telephony > Sets > All DNs**)
- Last Number Redial (dedicated key on the 2002/2004 models)
- Inbox Key (on 2002/2004 models)

The following table provides a short list of features available to IP phones in Local Mode. Only basic features such as Hold, Transfer, Call Forward, Last Number Redial, Intercom/DN key, and Inbox key can be programmed.

Table 14 Local mode features based on provisions in the IP Phone

SL No.	Feature code	Feature name	IP Phone type					
			2001, 1110, 2033, WLAN 2210, 2211, 2212, 6120, and 6140	2002, 1120E	2004, 2007, 2050, 1140E	1210	1220	1230
1	*79	Hold	Yes	Yes	Yes	Yes	Yes	Yes
2	*3	Transfer (Blind Transfer)	No	Yes	Yes	No	Yes	Yes
3	*4	Call Forward	No	Yes	Yes	No	Yes	Yes
4	*5	Last Number Redial	No	Yes	Yes	No	Yes	Yes
5	—	Intercom/ DN key	No	Yes	Yes	No	Yes	Yes
6	—	Line Keys	No	Yes	Yes	No	Yes	Yes

Nortel does not recommend the Button Programming action because the feature filtering options in effect in Local Mode make the Button Programming done in Element Manager useless.

For a list of Button Programming features, see *Nortel Business Communications Manager 5.0 Configuration —Devices*.

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 SRG. These feature settings are configured with the **Feature Settings** panel on the Business 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, 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 SRG, the *Nortel Business Communications Manager 5.0 Configuration—Telephony* 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 SRG is routed back to the SRG's local PSTN. Emergency Services Access (ESA) configuration on page 41 includes a procedure for configuring the SRG 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 SRG and the main office. This chapter details the procedures for establishing appropriate WAN connections to enable a VoIP network between the main office and SRG branch locations. Before proceeding, ensure that IP networking from the SRG to the WAN, and from the main office call server to the WAN have been configured and tested (see the figure IP networking, SRG to WAN, main office to WAN on page 63). SRG-specific configuration establishes the VoIP network (see the figure VoIP trunking configuration on page 69).

Figure 10 IP networking, SRG to WAN, main office to WAN

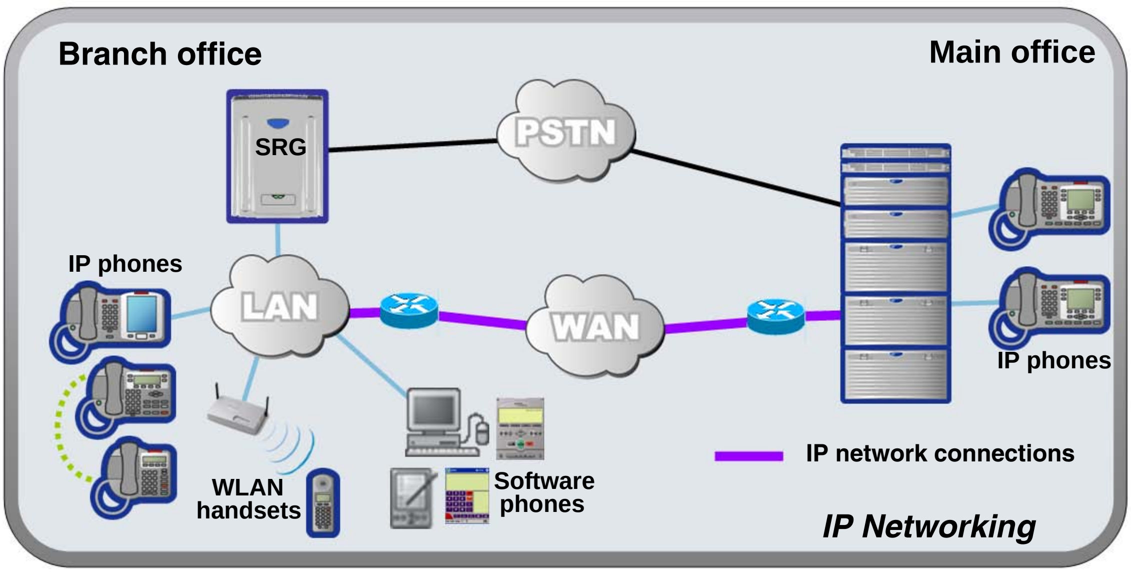
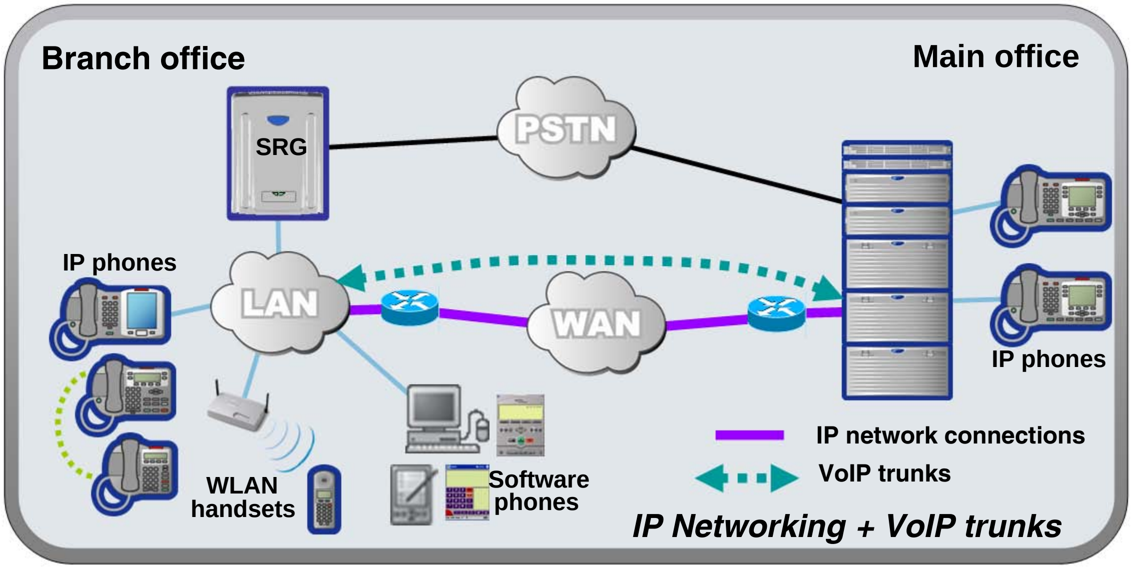


Figure 11 VoIP path for SRG operation



Generic procedures for setting up a private network on the SRG are covered in the *Nortel Business Communications Manager 5.0 Configuration—Telephony*. Items to address when establishing the private VoIP network between the SRG and the main office are:

- Basic parameters on page 65
- Private dialing plan on page 66
- Meridian Customer Defined Network (MCDN) on page 67
- QoS settings (codec, jitter buffer, and related items) on page 68
- Network security on page 69
- VoIP trunking configuration on page 69
- Line pools on page 72
- Alternate NRS for SIP trunking on page 72
- Call routing on page 73
- Outgoing calls configuration on page 73
- SRG PSTN access on page 75
- Main office information on page 77
- External attendant support on page 77
- Calling line ID support for PSTN to Communication Server 1000 tandem calls through SRG on page 77
- Support for paging facility in SRG on page 78
- Music during On-hold in SRG on page 79
- Modem over VoIP between SRG and Communication Server 1000 on page 79

Basic parameters

The table Basic parameters on page 65 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 *SRG50 5.0 Configuration Guide*.

Table 15 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 SRG. 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 SRG 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 SRG 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. See the *Networking Configuration Guide*.
DN range		Configured as part of BCM50 foundation configuration. DN range of the SRG 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 dialog box > Start DN field
Destination code for VoIP trunks		Configured for advanced routing. See Outgoing calls configuration on page 73 VoIP destination code = VOIP Trunk Access Code* = AC1** * VOIP Trunk Access Code is entered on the main office settings panel. See the server-specific chapters for details. ** For a UDP dialing plan, AC1 is the access code in the digit string <AC1> <LOC> <DN>.
SRG IP address		Configured as part of BCM50 foundation configuration. Configuration > System > IP Subsystem > IP Subsystem panel > General Settings tab > IP Settings details subpanel

Table 15 Basic parameters (Sheet 2 of 2)

Parameter	Value	Context
SRG net mask		Configured as part of BCM50 foundation configuration. Configuration > System > IP Subsystem > IP Subsystem panel > General Settings tab > IP Settings details subpanel
IP address of SRG gateway		Configured as part of BCM50 foundation configuration. Configuration > System > IP Subsystem > IP Subsystem panel > General Settings tab > IP Settings details subpanel
VLAN		If the SRG 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 SRG when in local mode		Required to specify a PSTN fallback route. See Outgoing calls configuration on page 73.
PSTN number for dialing into the SRG from the main office when in local mode		Required for main office configuration.

Private dialing plan

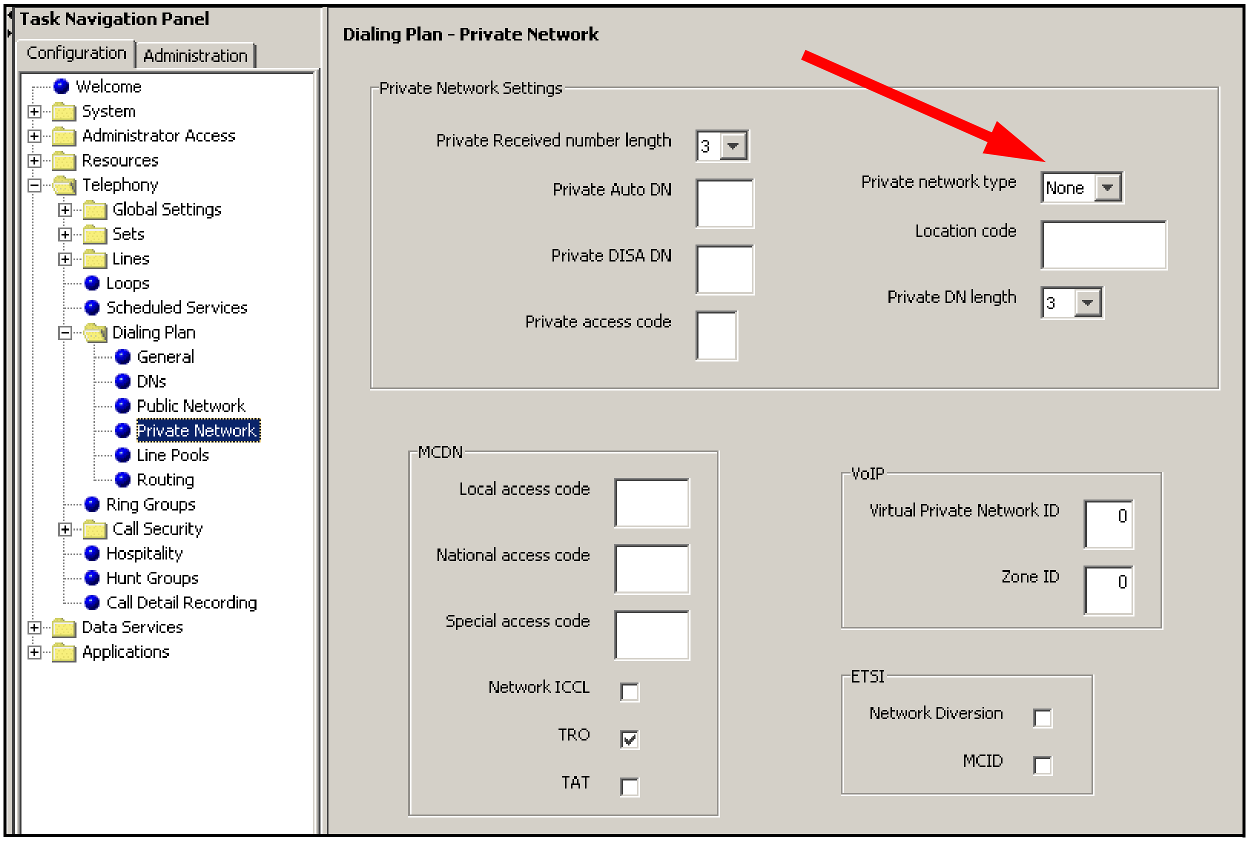
For SRG operation, either a coordinated dialing plan (CDP) or a uniform dialing plan (UDP) can be configured. Nortel recommends CDP because it requires the least dialing manipulation between the SRG and the main office call server. The dialing plan choice also determines whether the DN on the SRG matches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

Dialing plans between the SRG and the main office call server must be compatible. Private dialing plan configuration is described in detail in the *Nortel Business Communications Manager 5.0 Configuration—Telephony*.

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

The path to the Business 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 (see the figure Dialing Plan — Private Network panel, Private Network Settings subpanel on page 67).

Figure 12 Dialing Plan — Private Network panel, Private Network Settings subpanel



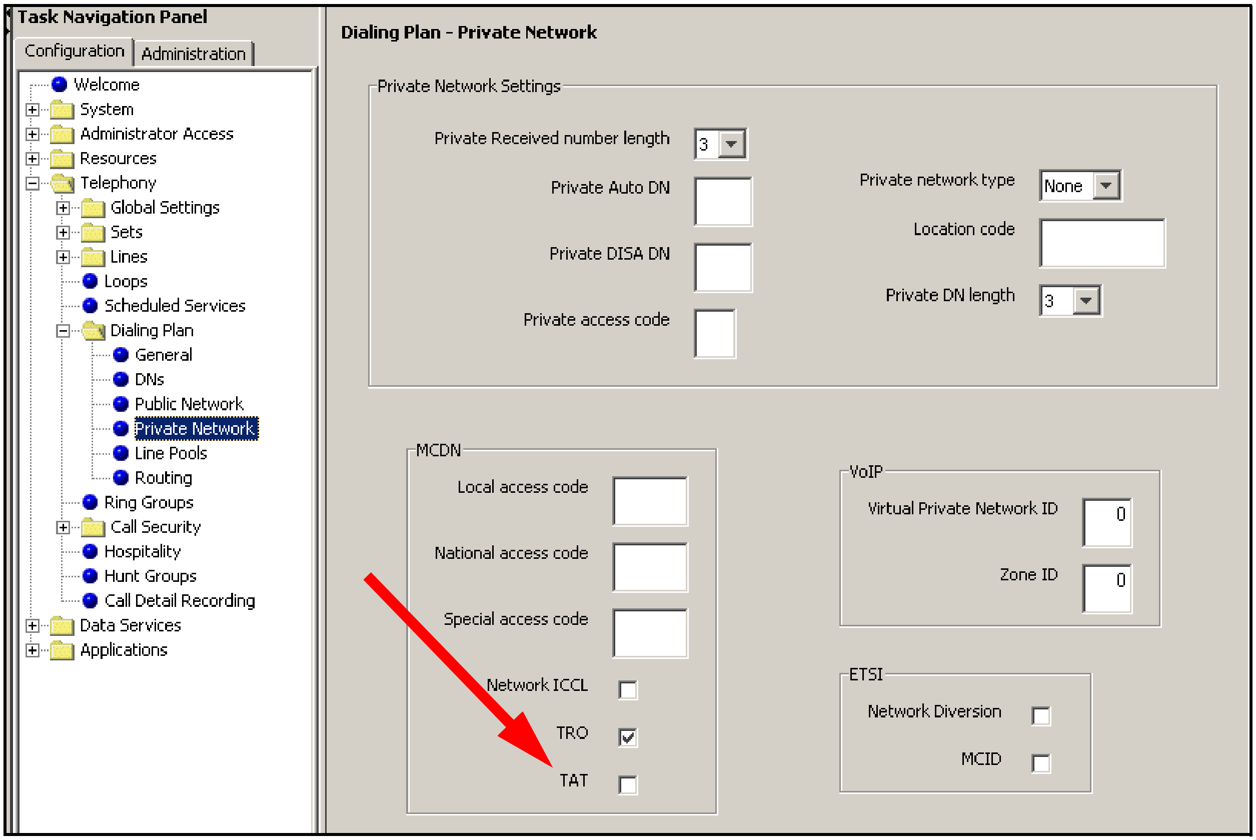
Meridian Customer Defined Network (MCDN)

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

To enable MCDN TAT

Use Business Element Manager to enable TAT by selecting the TAT checkbox at Configuration > Telephony > Dialing Plan > Private Network > Dialing Plan - Private Network panel > MCDN subpanel (see the figure Dialing Plan — Private Network panel, MCDN subpanel on page 68).

Figure 13 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 SRG are determined by the main office settings; the SRG settings must match the main office. Use the table Main office QoS settings on page 68 to record the main office settings, to facilitate configuration, and to provide a record of the datafill.

Table 16 Main office QoS settings (Sheet 1 of 2)

Media parameter	**H.323 settings**	**SIP settings**
Codec Preferences		
Silence compression (yes/no)		
Jitter buffer		
G.729 payload size (ms)		
G.723 payload size (ms)	Supported for IP telephones for local calls only. See IP telephone codec and jitter settings on page 53 for details.	

Table 16 Main office QoS settings (Sheet 2 of 2)

Media parameter	H.323 settings	SIP settings
G.711 payload size (ms)		
Incremental payload size (yes/no)		
T.38 fax (yes/no)		
Force G.711 for 3.1k audio (yes/no)		

In Business Element Manager, enter QoS settings for VoIP trunks through the Telephony Resources panel.

To enter the QoS settings for VoIP trunks

1 In Business Element Manager, navigate to **Configuration > Resources > Telephony Resources**.

2 On the **Modules** panel, locate the **Module type** column and select the **IP Trunks** row (see the figure Network security on page 69).

3 On the **Details for Module** subpanel, select the **H323 Media Parameters** tab or the **SIP Media Parameters** tab depending on which media parameters you want to configure.

4 Refer to the main office QoS settings recorded in the table Main office QoS settings on page 68 and enter the appropriate values.

Network security

Firewall configuration for SRG is the same as for the BCM50 and is detailed in the *Nortel Business Communications Manager 5.0 Configuration—Telephony*. Firewalls cannot be configured to allow VoIP pass through. Instead, the SRG supports IPsec tunnels to provide VoIP pass through. IPsec tunnels are also covered in the *Nortel Business Communications Manager 5.0 Configuration—Telephony*.

VoIP trunking configuration

When the SRG is operating in normal mode, connectivity to the main office call server is over VoIP trunks. An SRG can support up to 60 VoIP trunks.

Configuring VoIP trunks has three components:

- Fallback configuration on page 70
- Gatekeeper routing on page 71
- Line pools on page 72

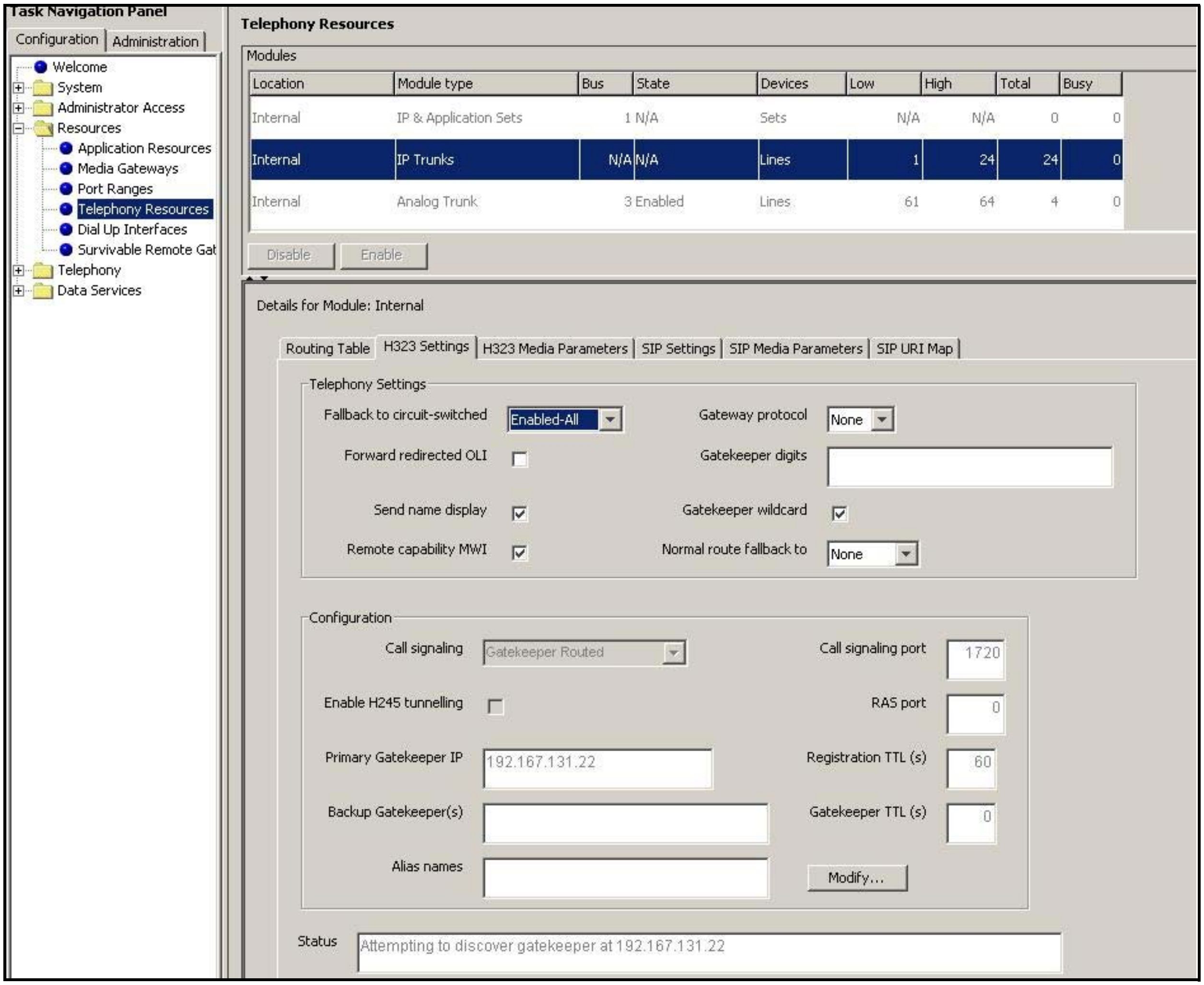
Fallback configuration

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

To enable fallback

1. Access the IP Trunks configuration panel in Business Element Manager (**Configuration > Resources > Telephony Resources > Modules** panel **> Module type** column: select **IP Trunks** (see the figure IP Trunks panel, H.323 Settings or SIP Settings on page 71).
2. On the Details for Module panel, select the **H323 Settings** tab or the **SIP Settings** tab.
3. On the Telephony Settings subpanel, locate the **Fallback to circuit-switched** field and use the list 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 **Enabled-TDM** if you want all TDM calls to be able to fallback to PSTN trunks if connectivity to the main office is lost.
   - c select **Disabled** if fallback is not required.

Figure 14 IP Trunks panel, H.323 Settings or SIP Settings



Gatekeeper routing

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 SRG and the SRG PSTN.

Examples:

- If the system is running with a CDP dialing plan and the SRG 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 SRG.
- If the PSTN to which the SRG connects has a location code of 521, the gatekeeper must have a record of this code in the SRG list so that main office calls to the SRG PSTN can be routed correctly.

If you require H323 trunks, you must select the Gatekeeper wildcard check box in the H323 settings. If you do not select the Gatekeeper wildcard check box, then calls are SIP routed (default setting).

When you select the Gatekeeper wildcard check box, all dialed digits match gatekeeper digits and VoIP calls are routed through the gatekeeper.

To configure gatekeeper settings

1. Access the IP Trunks configuration panel in Business Element Manager (Configuration > Resources > Telephony Resources > Modules panel > Module type column) select **IP Trunks** (see the figure IP Trunks panel, H.323 Settings or SIP Settings on page 71).
2. On the Details for Module panel, select the **H323 Settings** tab.
3. On the Telephony Settings subpanel, select the **Gatekeeper wildcard** check box.
4. On the Configuration subpanel, click **Modify**.
5. From the Call signaling list, select **Gatekeeper Routed**.
6. In the Primary Gatekeeper IP field, enter the IP address for the Primary Network Routing Services Address*
7. In the Backup Gatekeeper(s) field, enter the IP address for Alternate Network Routing Services Address*
8. In the Alias names field, enter **Name:** followed by the H.323 ID of the SRG (for naming conventions, see the *Nortel Business Communications Manager 5.0 Configuration—Telephony*).
9. Click **Ok**.
10. On the Telephony Setting subpanel, from the Gateway protocol list, select **CSE**.

* See CS 1000 information for the SRG on page 42.

Alternate NRS for SIP trunking

SRG 5.0 supports up to four SIP Proxies, and supports SIP Proxy failover. You can configure SIP Proxy to route all calls using proxy. SIP Proxy failover uses the SIP Proxy and the outbound proxy table configuration. The SIP Proxy domain is mandatory and is used in SIP message headers. For more information about SIP Proxy, see *Nortel Business Communications Manager 5.0 Configuration—Telephony* and *BCM 5.0 System Overview*.

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 public 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 *Nortel Business Communications Manager 5.0 Configuration—Telephony*.

Call routing

Call routing is covered in depth in the *Nortel Business Communications Manager 5.0 Configuration—Telephony*. The instructions in the *SRG50 5.0 Configuration Guide* are an abbreviation of that material, and only cover procedures that are specific to SRG operations; that is, for calls from redirected IP telephones. For more detailed information, see the *Nortel Business Communications Manager 5.0 Configuration—Telephony,* and *Nortel Business Communications Manager 5.0 Planning and Engineering.*.

Note: The DNs for the main office telephones system are marked off by the vacant number routing feature. SRG does not support Vacant Number Routing (VNR).

Instead, SRG uses Call Forward All Calls to emulate VNR for the SRG 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 SRG to route all the calls that are not terminated locally by the SRG. 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.

Outgoing calls configuration

To configure routing for outgoing calls

1 Create a schedule.

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

  b Select a **Schedule** (Sched 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 In the Services subpanel, select your schedule.

  f In the Routing Svc row for your schedule, select **Auto** from the list.

  g Select the **Overflow** checkbox for your schedule.

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).

  b Select the **Routes** tab.

  c Click **Add** to add a new route.

- **d** In the Route field, enter a number for the new route (for example, 998).
- **e** Click **Ok**.
- **f** Ensure that the DN Type is Public (Unknown).
- **g** Assign the VoIP line pool to the route (select the line pool from the **Use Pool** list; default is BlocA).
- **h** Click **Add** to add another new route.
- **i** In the Route field, enter a number for the new route (for example, 999).
- **j** Click **Ok**.
- **k** Ensure that the DN Type is Public (Unknown).
- **l** In the External Number field, enter the **PSTN number** of the main office.
- **m** Assign the PSTN line pool to the route (select the line pool from the **Use Pool** list; 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 SRG site.

- **a** Access the **Dialing Plan - Routing** panel (Configuration > Telephony > Dialing Plan > Routing).
- **b** Select the **Destination Codes** tab.
- **c** Click **Add** to add a new destination code.
- **d** In the Destination Code field, enter the new destination code (for example, 678).
- **e** Click **Ok**.
- **f** With the Destination Code row highlighted, select the **SRG** schedule from the Alternate Routes list (Alternate Routes for Destination Code details panel).
- **g** In the First Route field, enter **998** (the VoIP route).
- **h** In the adjacent Absorbed Length field, select **All** or **0** as 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.
- **i** In the Second Route field, enter **999** (the PSTN route).
- **j** 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 SRG 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.

SRG PSTN access

Access to the SRG PSTN is required for:

- calls to the SRG PSTN from SRG telephones or redirected SRG IP telephones in local mode
- SRG PSTN acess is needed as a fallback route in case of WAN failure, for calls from SRG Local mode sets and for inbound tandem calls from PSTN.
- calls from the SRG PSTN to redirected SRG IP telephones in normal mode
- calls from main office telephones to the SRG 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 SRG 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 SRG PSTN, or to redirect SRG IP telephones to use the SRG 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. LocateSet 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 > Trunk Type column > Details for Line panel > Restrictions tab > Use remote package field).

PSTN destination codes configuration

To allow SRG telephones to dial out over the PSTN and to allow main office telephones to tandem out through the local SRG 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).
2. Select the **Routes** tab. See the *Nortel Business Communications Manager 5.0 Configuration—Telephony* to:
   a. Add a new route (for the PSTN line pool).
   b. From the DN Type list, select **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** list).
3. Access the Dialing Plan - Routing panel (Configuration > Telephony > Dialing Plan > Routing).
4. Select the **Destination Codes** tab. See the *Networking Configuration Guide* to:
   a. Add the destination code to be used to access the local (SRG) PSTN.

      Users on both SRG and main office telephones dial this destination code to access the local (SRG) PSTN. If this code goes only to the SRG 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 SRG for SRG IP telephone calls that are connecting to the SRG PSTN.

      > **Note:** For main office programming, this code is the offnet dialing code that the gatekeeper recognizes for routing to the SRG.
      >
      > At the main office, zone-based digit manipulation is used to add a Zone Digit Prefix (ZDP) to PSTN calls from SRG 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 SRG IP telephones (to the SRG PSTN). The main office administrator f supplies this ZDP with the prerequisite information.

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

Main office information

The SRG requires information about the main office call server that is not needed for a BCM50. Business 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 CS 1000 considerations on page 31.

External attendant support

The SRG 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 SRG 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 SRG.
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 SRG does not have local attendant capability.

Calling line ID support for PSTN to Communication Server 1000 tandem calls through SRG

You can configure an SRG to tandem all incoming PSTN trunks to VoIP trunks. While operating in normal mode, the SRG directs these trunks to the Communication Server 1000. After configuration, the SRG processes the incoming calls on arrival, and establishes another VoIP call to the Communication Server 1000. The caller information (that is, calling number or name) received from the PSTN is passed as part of the VoIP call setup to the Communication Server 1000 for subsequent call processing (for example, to display on IP telephones).

Based on the type of the PSTN trunks, the following two conditions apply.

- After the PSTN trunks connects to the ISDN-based SRG (that is, PRI, BRI, or MCDN), the SRG receives the calling number or name as an out-of-band message during the time at which the VoIP call is established to the Communication Server 1000.

- After the PSTN connects in an analog trunk mode, the PSTN sends the calling number or name information between the first and second ring cycles (as defined by CLASS/CMS specifications) because the SRG processes incoming calls immediately on arrival from the PSTN. This information is never received by the SRG, and the SRG cannot relay this information to the Communication Server 1000.

This PSTN network limitation impacts the effectiveness of the SRG configuration with analog PSTN trunks by preventing the called party or Communication Server 1000 application from receiving calling number or name information. As the PSTN requires completion of a full ring cycle before sending the calling number or name on analog trunks, the SRG delays the immediate processing of incoming calls on analog trunks until the SRG receives the information.

Perform the following tasks to process the incoming calls on an analog trunk:

1. Assign an analog line to an analog set interface on the SRG.
2. Configure the analog set interface for Call Forward No Answer (CFNA) to the appropriate directory number on the Communication Server 1000, with a minimum forward delay of two ring cycles.

   This interval captures the incoming calling number or name information from the PSTN between the first and second ring cycle. The SRG passes the information to the Communication Server 1000 as part of the VoIP call setup.
3. Assign all incoming analog lines to a single analog set appearance.

   Connection of the physical analog set to the line is not needed.

Note: This configuration results in the caller hearing more rings before an answer versus the configuration where no calling number or name information is received. The Call Forward No Answer (CFNA) feature introduces an additional delay of two ring cycles to ensure the calling number or name is delivered from the PSTN. The caller hears another ring cycle prior to the SRG forwarding the call to the CS1000 as the caller ring back, and the ringing on the analog trunk is not synchronized. While the SRG redirects the call to the appropriate CS1000 user, additional ring cycles are generated on the CS1000.

Support for paging facility in SRG

Perform the following tasks to support paging on the SRG50.

1. Program a loop start analog trunk line with ACOD (for example, 88).
2. Connect the SRG50 trunk to an appropriate compatible paging device.
3. From a local SRG50 telephone, dial the ACOD to the paging device and confirm that the connection is established.

Perform the following tasks to support paging on the Communication Server 1000.

1. Ensure that there are VoIP virtual trunks setup between the main and the SRG50 site.
2. Ensure that the NRS on the local signalling server is turned on.
3. Ensure that the CDP is enabled.
4. Build the new RLI, which is routed over VoIP trunks to the SRG site.
5. Program the ACOD (for example, 88) in the Communication Server 1000 as a Trunk Steering Code (TSC) and route to the new RLI created in step 4.
6. In NRS, ensure that the SRG50 is defined as an endpoint. If not, define the SRG50 as a static endpoint (SRG IP address).
7. In NRS, define ACOD (for example, 88) as a routing entry pointed to the SRG50 system.
8. Test the call from the IP set located at the SRG50 site.

Music during On-hold in SRG

During the local mode, on-hold is available for incoming calls received over a PSTN trunk. The on-hold supports three options; Silence, Tones, and Music.

Selecting an On-hold menu

1. Click **Configuration** > **Telephony** > **Global Settings** > **Feature Settings** > **On hold.**
2. Select the option that you want from the list.

> **Note:** The external music source supports the music option that is attached to the SRG50 music source input jack. For more information, see *Installation Checklist and Quick Start Guide.*

Modem over VoIP between SRG and Communication Server 1000

The Communication Server 1000 supports modem-over-IP interworking.

The typical configuration is: PC (Hyper terminal)<--->analog<--->Modem<--->analog port/ SRG50 R3<--->sip trunk<--->CS1000 Rls 5.5/analog port<--->Modem<--->analog<--->PC (Hyper terminal).

Table 17 Modem call matrix

Config scenario number	Codec set in SRG	Codec set in CS1000	Call direction	Force G711 flag within SRG	Codec selected for the modem call	Comments
1	G.711	G.729	SRG--->CS1K	enabled	**G.711**	—
2	G.711	G.729	SRG--->CS1K	disabled	**G.711**	**Failed** as CS1000 not forced to use G.711
3	G.711	G.729	CS1K--->SRG	enabled	**G.711**	—
4	G.711	G.729	CS1K--->SRG	disabled	**G.711**	**Failed** as CS1000 not forced to use G.711
5	G.729	G.711	SRG--->CS1K	enabled	**G.711**	—
6	G.729	G.711	SRG--->CS1K	disabled	**G.711**	**Failed** as CS1000 not forced to use G.711
7	G.729	G.711	CS1K--->SRG	enabled	**G.711**	—
8	G.729	G.711	CS1K--->SRG	disabled	**G.711**	—

Chapter 7
PSTN access and analog devices

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

- PSTN access considerations on page 81
- Analog devices considerations on page 82

PSTN access considerations

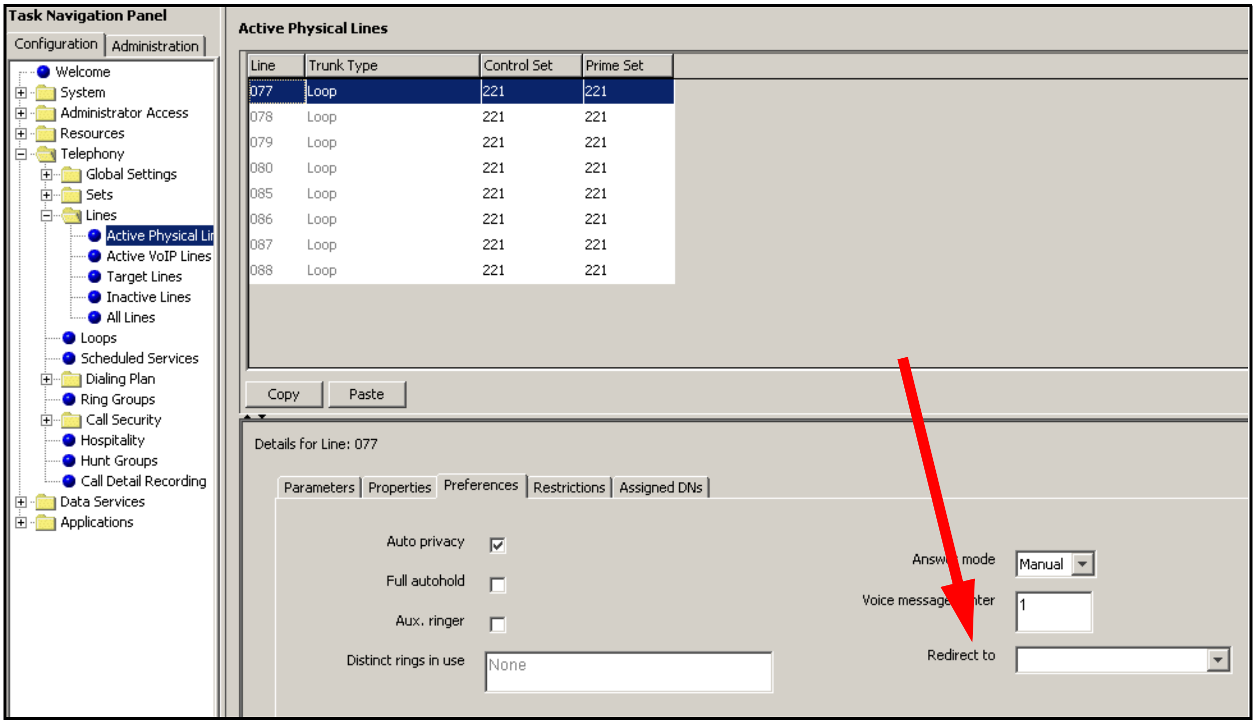
Consider the following for PSTN access:

- **PSTN access**
  To provide access to the SRG PSTN when the SRG is in local mode, or to be able to set up tandem dialing from the main office through the SRG to the SRG PSTN, one or more PSTN trunks must be configured on the system. See the *Networking Configuration Guide*.
- **Tandem calls**
  The SRG 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 SRG PSTN, or to allow redirected SRG IP telephones to use the SRG local PSTN, a remote access package must be specified to provide access to the PSTN line pool. This procedure is covered in Outgoing calls configuration on page 73.
- **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). See the figure Configuring manual and auto-answer lines for SRG operation on page 82.

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

Figure 15 Configuring manual and auto-answer lines for SRG operation



Analog devices considerations

Consider the following for analog devices:

- **Basic operation**
  Analog telephones and devices connected to the SRG always function as local telephones to the SRG. 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.
- **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 SRG 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. In Business Element Manager, navigate to the Active Sets panel (Configuration > Telephony > Sets > Active Sets).
2. Select the **Line Access** tab.

3 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.

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

5 To assign specific PSTN lines to each telephone, add the line(s) (Details for DN 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.

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

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

8 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.

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

10 From the Details for DNs panel, select the **Capabilities** tab.

11 From the Handsfree list, select **None**.

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

13 Clear the **Paging** check box.

14 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 on page 85
- IP terminal details on page 86
- Probable causes for redirection failure on page 87
- Troubleshooting fallback to local mode on page 88
- IP telephones manual redirection on page 88

IP telephone troubleshooting

The table IP telephone troubleshooting on page 85 provides issues and solutions for IP Phone troubleshooting.

Table 18 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 SRG. Double check the IP configuration of the telephone and the IP connectivity to the SRG (cables, switches, and so on).
Slow connection between the handset and the Business Communications Manager	If the connection between the IP client and the SRG 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 18 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 SRG 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 SRG 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 table IP Terminal Details on page 86 summarizes the events that can be raised by the SRG. The events and details appear in Business Element Manager at Configuration > 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 SRG state machine uses flags to indicate that a given event has been logged.

Table 19 IP Terminal Details (Sheet 1 of 2)

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

Table 19 IP Terminal Details (Sheet 2 of 2)

Details	Event Id	Severity	Call Server Type	Comments
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 SRG.
SRG terminated unexpectedly.	57750	Critical	All	Indicates that a critical error caused the SRG 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 (see IP terminal details on page 86); the IP telephone remains registered with the SRG in local mode operation.

Definitive causes for registration failure depend on the main office call server. These causes can include:

- The registration password entered at the SRG 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.
- The actual IP telephone set type at the SRG 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.

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:

1 IP telephone firmware discrepancies

The SRG supports automatic firmware updates (see CS 1000 considerations on page 31). 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 SRG.

Check the IP Terminal Details tab (see IP terminal details on page 86) 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 SRG first, then on the main office equipment.

When the IP telephone firmware is updated on the main office, the main office redirects all SRG IP telephones back to the SRG for a firmware upgrade. If the SRG has already been patched with the new firmware, the telephone is upgraded when it registers with the SRG. 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 SRG has not been upgraded with the new firmware, the telephone is redirected back to the main office.

2 Gatekeeper failure

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 SRG and stays in local mode.

Troubleshoot the problem by checking the settings made when implementing the CS 1000 information for the SRG on page 42 and Fallback configuration on page 70. If you make changes, manually redirect the telephones (IP telephones manual redirection on page 88, below).

IP telephones manual redirection

To manually redirect an IP telephone to the main office

1. Access the Configuration > Telephony > Sets > Active Sets > IP Terminal Details.
2. Click 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. See IP terminal details on page 86.

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

For specific information about making calls and using features, see the feature guides for the main office application.

Appendix A
Telephone features in normal and local mode

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

The SRG50 supports the following:

- IP Phones 1100, 1120E, and 1140E
- IP Phones 1210, 1220, and 1230
- IP Phones 2001, 2002, 2004, and 2007
- IP Audio Conference Phone 2033
- IP Softphone 2050
- Mobile Voice Client (MVC) 2050
- WLAN Handsets 2210, 2211, 2212, 6120, and 6140
- Analog (500/2500 type) telephones
- Analog devices such as fax machines

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 SRG 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 SRG and depend on the SRG applications. Consult the SRG system administrator for a complete description.

A quick reference list to the default SRG features for an analog telephone are provided in ATA extension features on page 96. Consult the SRG administrator to determine if these features have been changed.

Local mode

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

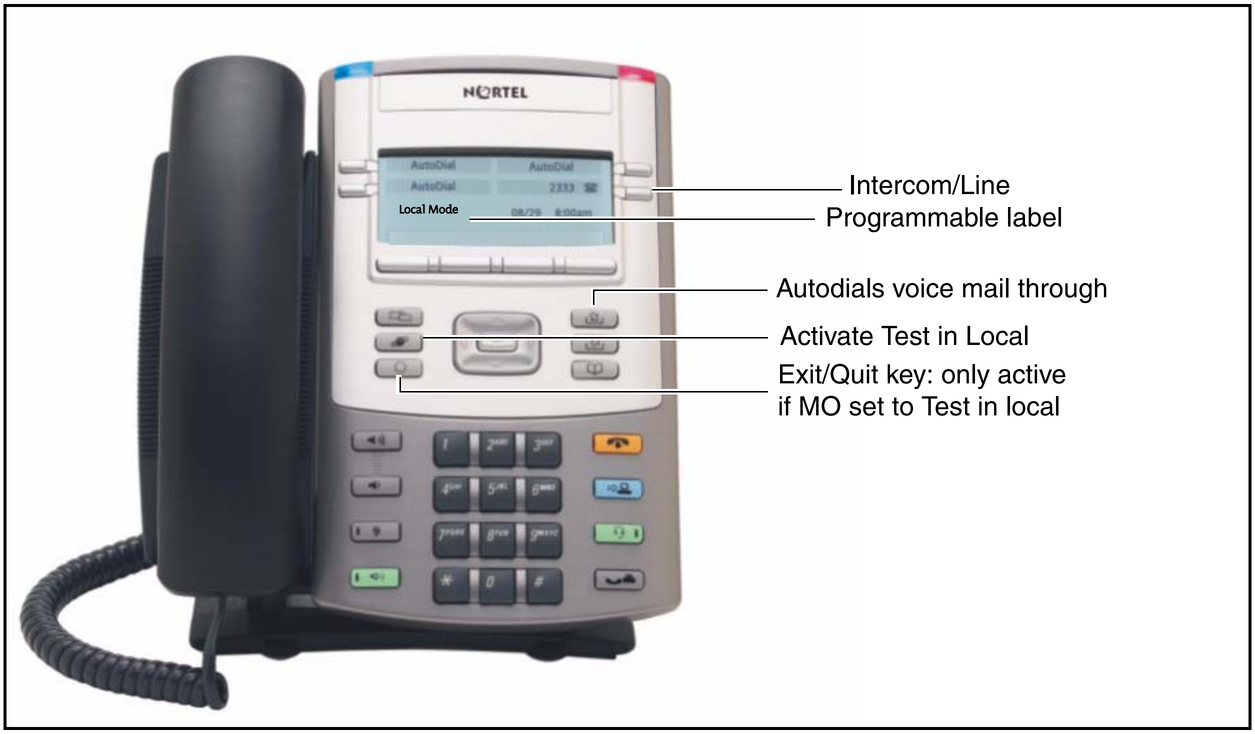
If routing and destination codes are set up as suggested in Call routing on page 73, the dialing sequence for the main office is the same as in normal mode.

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

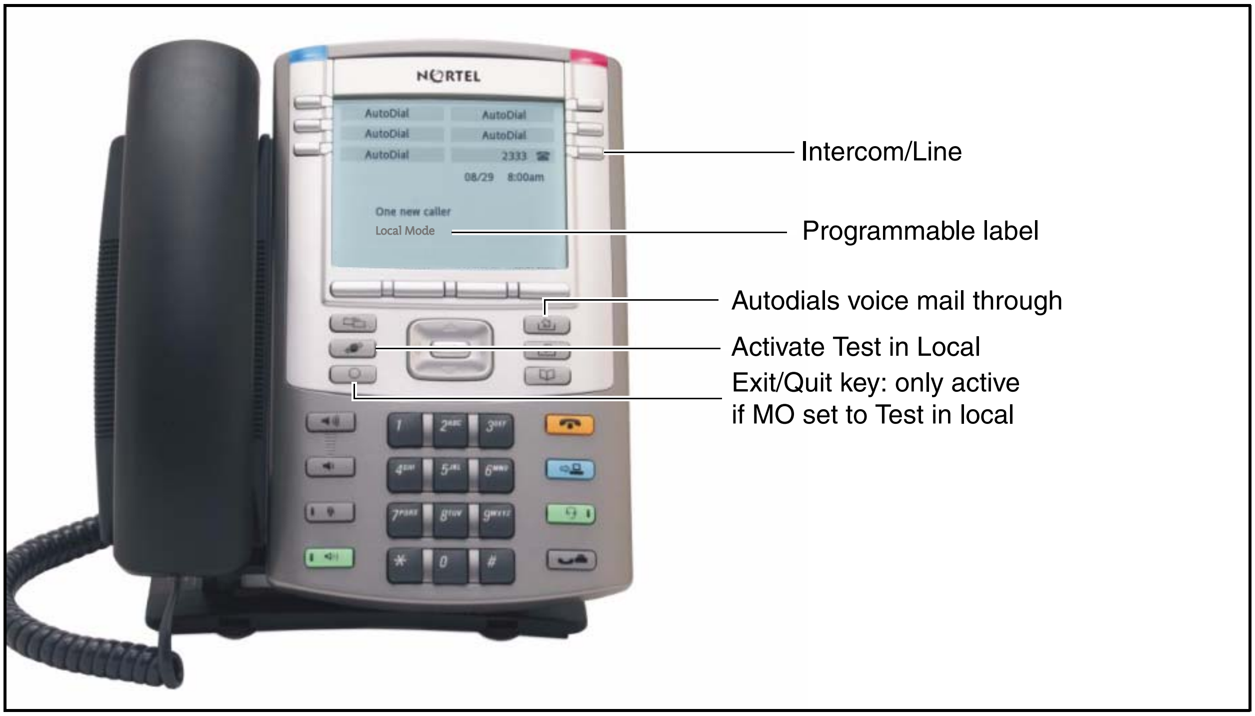
- IP Phone 1120E in Local mode on page 92
- IP Phone 1140E in Local mode on page 93
- IP Phone 2001 in Local mode on page 93

- IP Phone 2002 in Local mode on page 94
- IP Phone 2004 in Local mode on page 94
- IP Phone 2007 in Local mode on page 95
- IP Phone 2033 in Local mode on page 95
- IP 2050 Softphone in Local Mode on page 96

IP Phone 1120E in Local mode

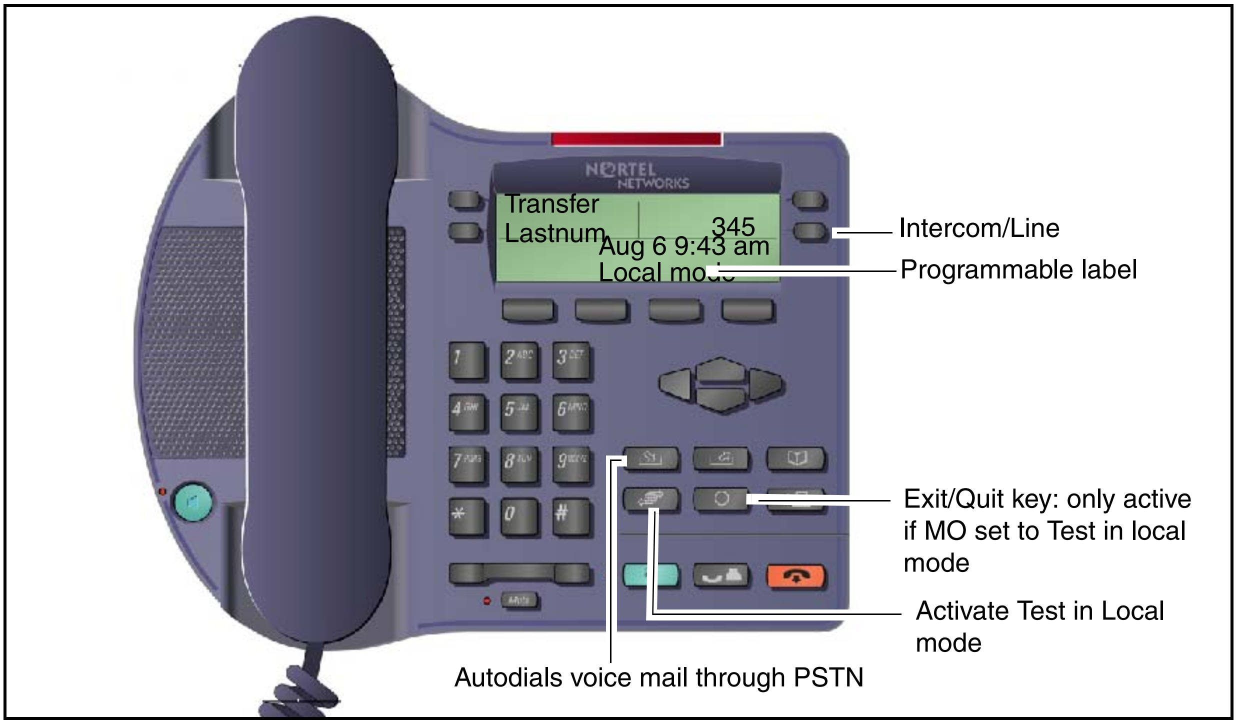


IP Phone 1140E in Local mode

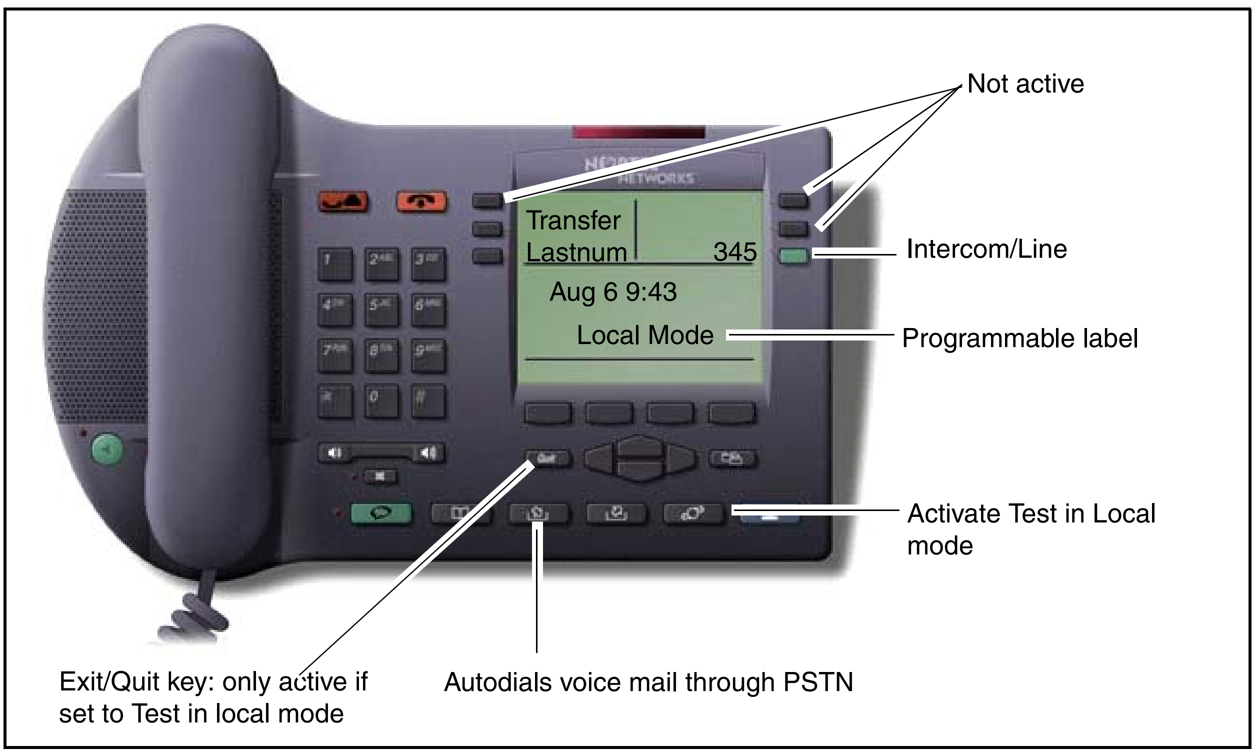


IP Phone 2001 in Local mode

IP Phone 2002 in Local mode



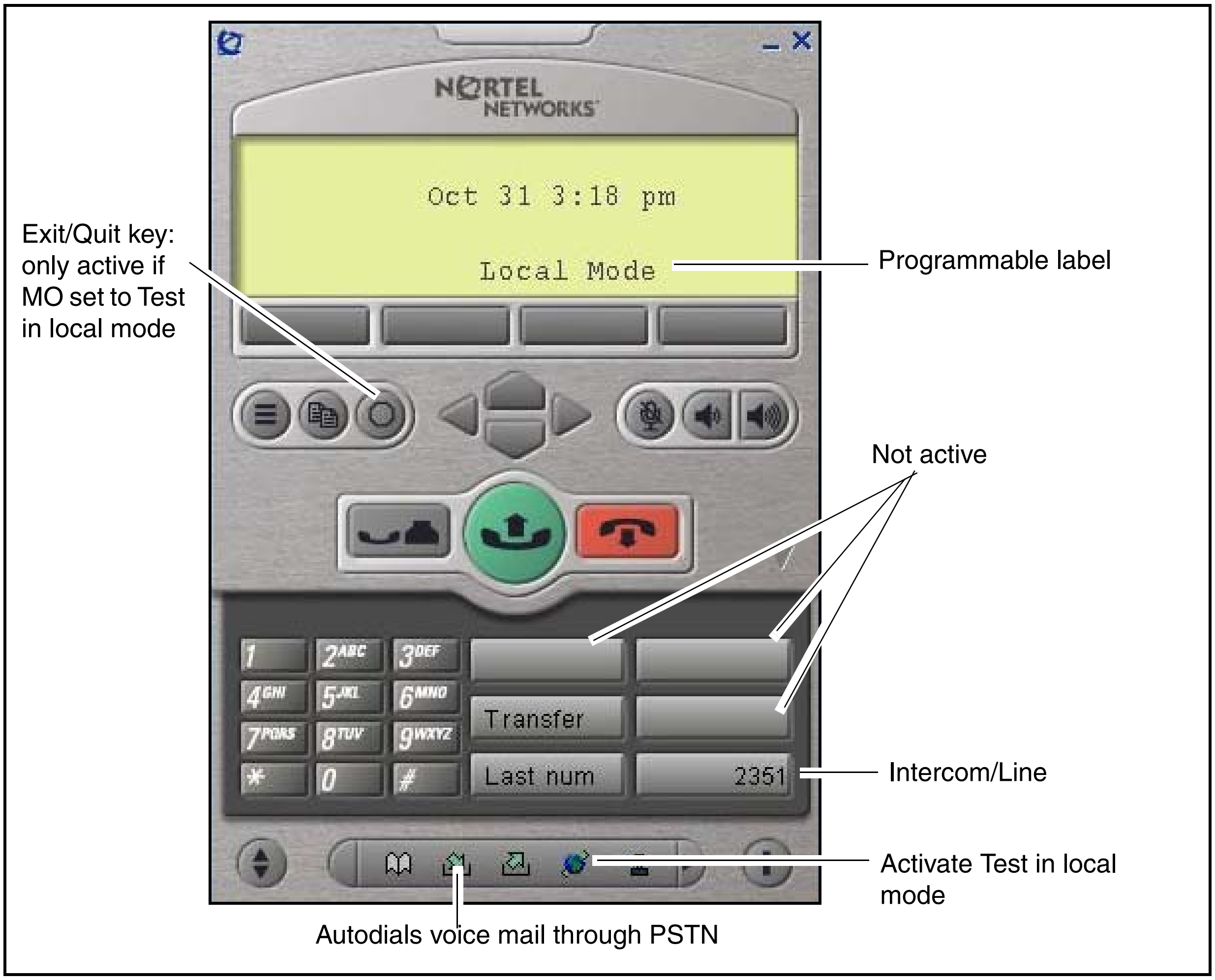
IP Phone 2004 in Local mode



IP Phone 2007 in Local mode

IP Phone 2033 in Local mode

IP 2050 Softphone 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. For specific key sequences, see the table Link key sequences on page 96.

Table 20 Link key sequences (Sheet 1 of 2)

Feature	Activate	Cancel		Feature	Activate	Cancel
Alternate line	LINK 2			Privacy control	LINK *83	
Call Forward (local system)	LINK *4	LINK #4		Link	LINK *71	

Table 20 Link key sequences (Sheet 2 of 2)

Feature	Activate	Cancel		Feature	Activate	Cancel
Call Forward (external system)	LINK *4 <dialed #> LINK 2	LINK #4		Pause	LINK *78	
				Timed release	LINK *72	
				Ring Again	LINK *2	LINK #2
Call parking	LINK *74			Saved Number Redial	LINK *67	
Call pick-up (Directed)	LINK *76					
Call pick-up (Group)	LINK *75			Tones	LINK *809	LINK #809
Call Queuing	LINK *801			Transfer	LINK *70	
Camp-on	LINK *82			Trunk Answer	LINK *800	
Conference call	LINK *3			Voice Call	LINK *66	
Hold Call (Exclusive)	LINK *79					
Hold Call (Public)	LINK 2			Voice messaging - Internal		
Last Number Redial	LINK *5			Access mailbox	LINK *981	
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 SRG supports CDP or UDP only. If the main office is running BDP, the SRG 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 SRG must be assigned to all gatekeepers on the network. If the gatekeeper is down, the SRG attempts to connect to the alternate gatekeeper, if there is one. If the alternate gatekeeper is also down, or there is no alternate gatekeeper, the SRG IP telephones remain registered with the main office, but calls cannot be sent to the SRG.

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 SRG 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 SRG IP telephones when connectivity with the main office is unavailable.

main office, main office call server
: The system that provides telephony services to redirected SRG 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 SRG 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.

SIP
: An IP gateway protocol used by both the main office and the SRG to create VoIP trunking connections.

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 SRG 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 SRG 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 SRG local-PSTN call dialed from an SRG IP telephone. The number differentiates the call from a main office local-PSTN call dialed by a main office telephone.

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