BCM 4.0 Networking Configuration Guide

BCM 4.0

Business Communications Manager

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Task List

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Chapter 1
Getting started with BCM

Refer to the following topics for general BCM information:

- "About BCM"
- "Symbols and conventions used in this guide" on page 31
- "Related publications" on page 32
- "How to get Help" on page 37

About this guide

The *BCM 4.0 Networking Configuration Guide* describes how to install, configure, and maintain the BCM200, BCM400, and BCM1000 hardware running Business Communications Manager 4.0 (BCM) 4.0 software.

Purpose

The concepts, operations, and tasks described in this guide relate to the hardware of the BCM system. This guide provides task-based information on how to configure a BCM network.

Use Element Manager, Startup Profile, and Telset Administration to configure various BCM parameters.

In brief, the information in this guide explains:

- public and private networking
- configuring trunks and lines
- media gateways
- loops
- IP settings
- dialing plans

Audience

The *BCM 4.0 Networking Configuration Guide* is directed to installers responsible for installing, configuring, and maintaining BCM systems.

To use this guide, you must:

- be an authorized BCM installer/administrator within your organization
- know basic Nortel BCM terminology
- be knowledgeable about telephony and IP networking technology

Acronyms

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

Table 1 Acronyms (Sheet 1 of 2)

Acronym	Description
AH	authentication header
ARP	address resolution protocol
ARS	automatic route selection
ASM	analog station module
ATA	analog terminal adapter
BCM	Business Communications Manager
BRI	basic rate interface
CbC	Call-by-call
CoS	Class of Service
CLID	calling line identification
CLIR	calling line information restriction
CRC	cyclic redundancy check
CSU	Channel Service Unit
DHCP	Dynamic host configuration protocol
DID	direct inward dial
DISA	direct inward system access
DLCI	data link connection identifier
DNS	domain name server
DTMF	dual tone multi-frequency
FEM	fiber expansion module
FoIP	fax over IP
MCDN	Meridian customer defined networking
MCID	malicious call identification
MSC	media services card
NAT	network address translation
OLI	outgoing line identification
ONN	outgoing name and number
OSPF	open shortest path first
PFS	perfect forward security
PPPoE	point to point over Ethernet
QoS	quality of service
RIP	routing information protocol
SIP	session initiated protocol

Table 1 Acronyms (Sheet 2 of 2)

Acronym	Description
SRG	survivable remote gateway
TAT	trunk anti-tromboning
TTL	time to live
UDP	universal dialing plan
VAD	voice activity detection
VLAN	virtual LAN
VoIP	voice over IP

Organization

This guide is organized for easy access to information that explains the concepts, operations, and procedures associated with the BCM system.

About BCM

The BCM system provides private network and telephony management capability to small and medium-sized businesses.

The BCM system:

- integrates voice and data capabilities, VoIP gateway functions, and QoS data-routing features into a single telephony system
- enables you to create and provide telephony applications for use in a business environment

BCM key hardware elements

BCM includes the following key elements:

- BCM200 main unit
- BCM400 main unit
- BCM1000 main unit
- BCM expansion unit (compatible with BCM400 main unit)
- BCM400 expansion gateway
- BCM media bay modules (MBM):
  - — 4x16
  - — ASM8, ASM8+
  - — BRIM
  - — CTM4, CTM8
  - — DDIM
  - — DSM16+, DSM32+

- — DTM
- — FEM
- — GASM
- — GATM4, GATM8

BCM features

BCM supports the complete range of IP telephony features offered by existing BCM products:

➡ **Note:** You enable the following features by entering the appropriate keycodes (no additional hardware is required).

- VoIP Gateway: Up to 12 VoIP trunks
- VoIP Telephony Clients: Up to 64 VoIP Telephony clients, supporting the range of Nortel IP Phones.

BCM applications

BCM supports many applications provided on the existing BCM platforms.

➡ **Note:** You enable the following features by entering the appropriate keycodes (no additional hardware is required).

- Voice Messaging for standard voice mail and auto-attendant features
- Unified Messaging providing integrated voice mail management between voice mail and common e-mail applications
- Fax Suite providing support for attached analog fax devices
- Voice Networking features
- LAN CTE (computer telephony engine)
- VEWAN (Voice Enabled WAN)
- IVR (Integrated Voice Response)
- IP Music
- Intelligent Contact Center

Symbols and conventions used in this guide

These symbols are used to highlight critical information for the BCM 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 a default should be changed, or where the administrator needs to make a decision about 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 BCM main unit and expansion unit power cords from the ac outlet before performing any maintenance procedure.

The following conventions and symbols are used to 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.
Underlined word in capital letters (shown in the bottom line of a two-line display telephone)	`PLAY`	Display option. Available 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.

The following text conventions are used in this guide to indicate the information described:

Convention	Description
`bold Courier text`	Indicates command names and 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 coordinating icon on whichever set you are using.

Related publications

This section provides a list of additional documents referred to in this guide. There are two types of publications: Technical Documents on page 32 and User Guides on page 34.

Technical Documents

BCM 4.0 System Overview (N0060607)

System Installation

BCM 3.x to BCM 4.0 Upgrade Guide (N0060597)

BCM 4.0 Installation Checklist and Quick Start Guide (N0060602)

BCM1000 BCM 3.7 Installation and Maintenance Guide (N0008587 01)

BCM 4.0 for BCM1000 Installation and Maintenance Guide Addendum (N0060603)

BCM200/400 BCM 4.0 Installation and Maintenance Guide (N0060612)

Keycode Installation Guide (N0060625)

BCM R2MFC Installation and Configuration Guide (N0027684*)*

Project Management Guide (N0060632)

System Programming

BCM 4.0 Administration Guide (N0060598)

BCM 4.0 Device Configuration Guide (N0060600)

BCM 4.0 Networking Configuration Guide (N0060606)

BCM 4.0 Telset Administration Guide (N0060610)

Telephones and Peripherals

BCM 4.0 Telephony Device Installation Guide (N0060609)

BST Doorphone Installation and Configuration Guide (P1013654)

T24 KIM Installation Card (P0603481)

IP Key Expansion Module (KEM) User Guide

Digital Mobility

DECT Deployment and Demonstration Tool

Digital Mobility System Installation and Configuration Guide (N0000623)

T7406 Cordless Handset Installation Guide (P0606142)

2G4 Deployment and Demonstration Tool (N0027187)

IP Telephony

i2050 Software Phone Installation Guide (N0022555)

IP Phone 1120E User Guide (NN-10300-062)

IP Phone 1140E User Guide (NN-10300-064)

IP Audio Conference Phone 2033 User Guide (N0060623)

WLAN IP Telephony Installation and Configuration Guide (N0060634)

Call Pilot

BCM 4.0 Unified Messaging Configuration Guide (N0060611)

CallPilot Fax Set Up and Operation Guide (P0606017)

CallPilot Manager Set Up and Operation Guide (N0027247)

CallPilot Message Networking Set Up and Operation Guide (N0027249)

CallPilot Programming Record (N0027404)

CallPilot Reference Guide (N0060617)

CallPilot Telephone Administration Guide (N0060618)

Contact Center

ipView Software Wallboard Set Up and Operation Guide (N0027284)

Contact Center Reports Explained (N0060635)

Contact Center Set Up and Operation Guide (N0060620)

Multimedia Contact Center Set Up and Operation Guide (N0060626)

Multimedia Contact Center Web Developer Guide (N0060627)

Reporting for Contact Center Set up and Operations Guide (N0060637)

Upgrading from Call Center Reporting to Reporting for Contact Center (N0060638)

CallPilot Contact Center Telephone Administration Guide (N0060615)

IVR

Media Processing Server Series COMMGR Reference Manual (P0988083)

PeriReporter User's Guide (P0988093)

PeriView Reference Manual (P0988094)

IVR Installation and Configuration Guide (N0060624)

BCM IVR Integration Supplement (P0995957)

Other Applications

BCM 4.0 LAN CTE Configuration Guide (N0060604)

BCM 4.0 Call Detail Recording System Administration Guide (N0060599)

Consolidated Reporting Data (N0064482)

BCM Imaging Tool User Guide (P0609711)

User Guides

Telephones and Peripherals

BCM 4.0 Telephone Features User Guide (N0060608)

BST Doorphone User Guide (P0605668*)*

Central Answering Position (CAP) User Guide (P0603480)

Hospitality Features Card (N0027326)

System-wide Call Appearance (SWCA) Features Card (N0027186)

T7000 Telephone User Card (P0912061)

T7100 Telephone User Card (P0609621)

T7208 Telephone User Card (P0609622)

T7316 Telephone User Card (P0935248)

T7316E Telephone User Card (P0609623)

Digital Mobility

DECT 413X/414X Handset User Guide (N0028550)

Digital Mobility Phone 7420 User Guide (N0000635)

Digital Mobility Phone 7430/7440 User Guide (N0028550)

T7406 Cordless Telephone User Card (P0942259)

IP Telephony

IP Audio Conference Phone 2033 User Guide (N0060623)

IP Phone 2001 User Guide (N0027313)

IP Phone 2002 User Guide (N0027300)

IP Phone 2004 User Guide (N0027284)

IP Phone 2007 User Guide (N0064498)

BCM WLAN 2210/2211/2212 Handset User Guide (N0009103)

Call Pilot

CallPilot 2.5 Unified Messaging User Guide for Internet Clients

CallPilot 2.5 Unified Messaging User Guide for Lotus Notes

CallPilot 2.5 Unified Messaging User Guide for Microsoft Outlook

CallPilot 2.5 Unified Messaging User Guide for Novell GroupWise

CallPilot Fax User Guide (N0027227)

CallPilot Message Networking User Guide (N0027253)

CallPilot Quick Reference Card - CP Interface (N0027401)

CallPilot Quick Reference Card - NVM Interface (N0027379)

CallPilot Quick Reference Card - Remote Users (CP Interface) (N0027359)

CallPilot Quick Reference Card - Remote Users (NVM Interface) (N0027346)

Contact Center

Contact Center Agent Guide (N0060619)

Contact Center Supervisor Guide (N0060621)

Other Applications

BCM 4.0 Personal Call Manager User Guide (N0027256 02)

CallPilot Message Networking User Guide (N0027253)

CallPilot Quick Reference Card - CP Interface (N0027401)

CallPilot Quick Reference Card - NVM Interface (N0027379)

CallPilot Quick Reference Card - Remote Users (CP Interface) (N0027359)

CallPilot Quick Reference Card - Remote Users (NVM Interface) (N0027346)

Contact Center

Contact Center Agent Guide (N0060619)

Contact Center Supervisor Guide (N0060621)

Other applications

BCM 4.0 Personal Call Manager User Guide (N0027256 02)

How to get Help

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

Getting Help from the Nortel Web site

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

http://www.nortel.com/support

This site enables customers to:

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

Getting Help over the phone from a Nortel Solutions Center

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

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

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

http://www.nortel.com/callus

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

http://www.nortel.com/erc

Getting Help through a Nortel distributor or reseller

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

Chapter 2
System telephony networking overview

The system supports both public and private networking for telephony traffic.

- The public network is created by PSTN trunk connections from a Central Office terminating on a telephone system such as the BCM.
- A private network is created when the system is connected through dedicated PSTN lines or VoIP trunks to other systems. This system may take several forms. At the simplest level, your system may be behind a private PBX, which connects directly to the Central Office. A more complicated system may be a node in a network of systems of various types, where calls not only terminate at the system, but calls may need to be passed through the system to other nodes unconnected to the originating node.

Refer to the following topics:

- "Basic system configurations"
- "Private network parameters" on page 43

Basic system configurations

In the most basic application, your system can provide support for system telephones to make and receive calls over public network (PSTN) lines.

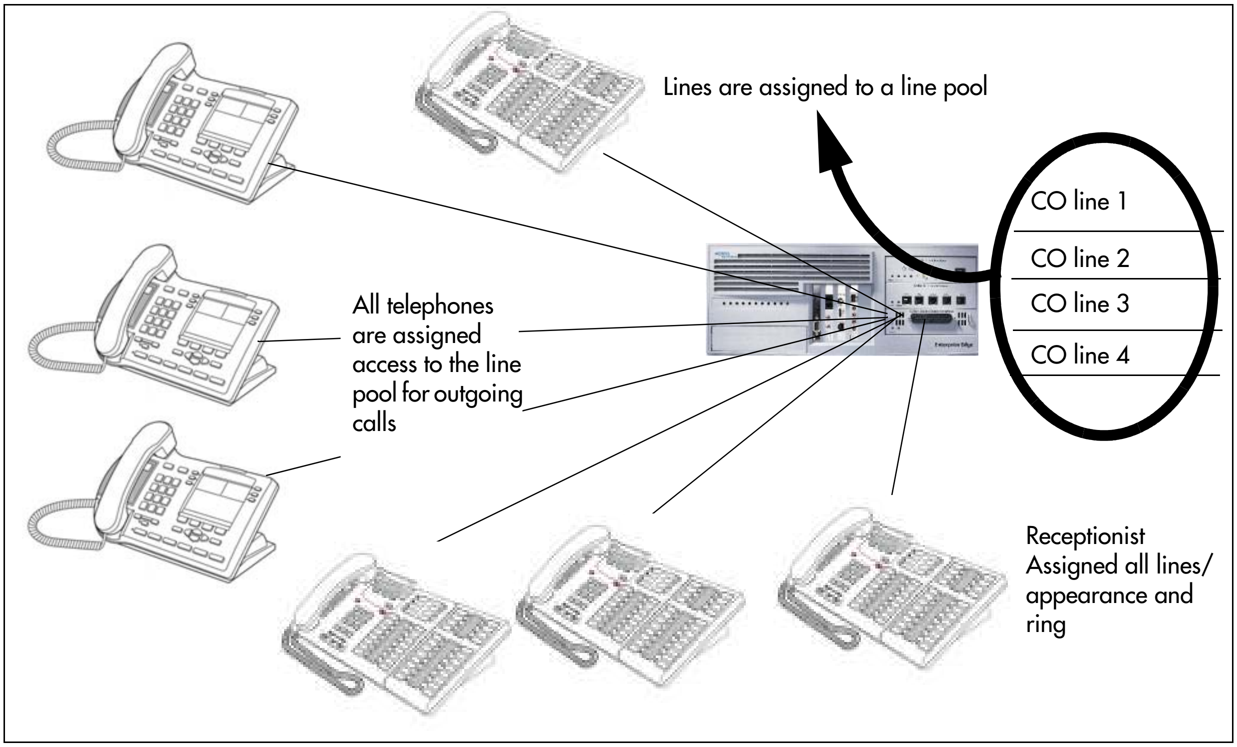
Two basic system telephony configurations

The following provides a broad overview of the telephony setup for a PBX and a DID system.

PBX system

This setup is for a larger offices which have fewer CO lines than there are telephones. In this case the lines are pooled, and the line pool access is assigned to all DNs. There may also be a designated attendant with a telephone that has all lines individually assigned.

Figure 1 PBX system


Lines are assigned to a line pool
CO line 1
CO line 2
CO line 3
CO line 4
All telephones are assigned access to the line pool for outgoing calls
Receptionist Assigned all lines/ appearance and ring

Incoming calls

1. A call comes in on a line.
2. The receptionist answers the call and finds out who the call is for.
3. The receptionist transfers the call to a specific telephone (DN).
4. The person can pick up the call at that DN only.

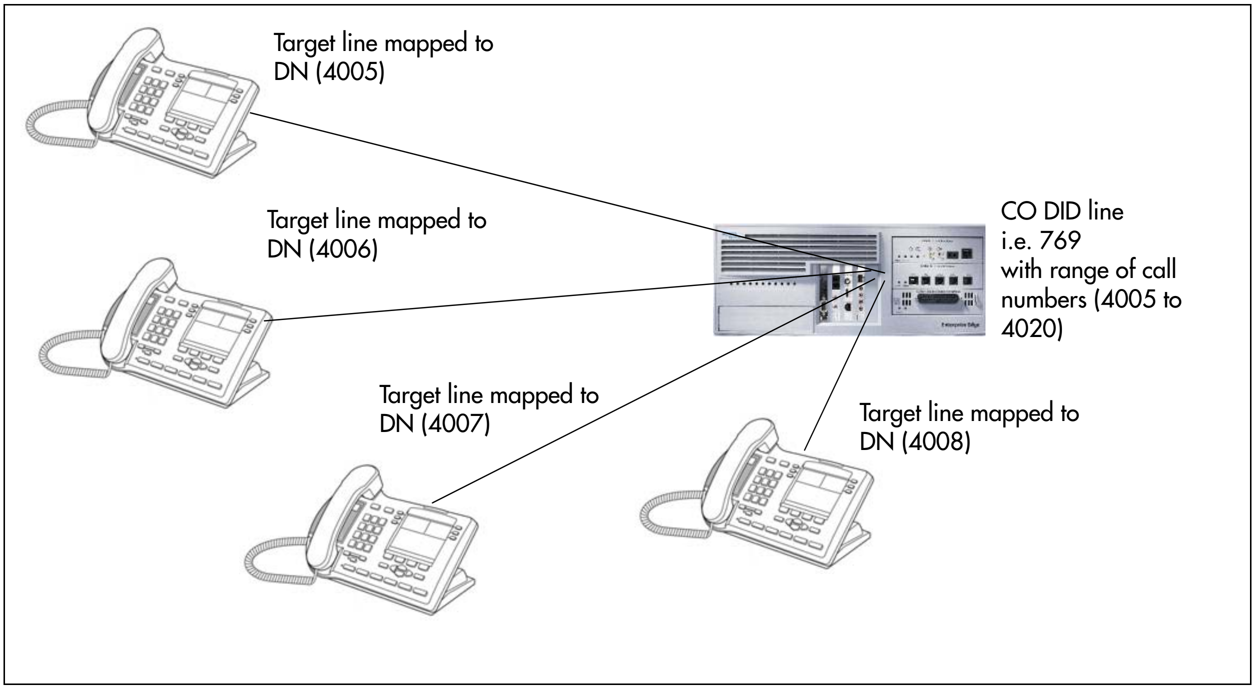
Outgoing calls

1. User selects the intercom button or dials a line pool access code, which selects a line in the line pool.
2. The user dials the outgoing telephone number.

DID system

This setup allows you to assign a dedicated phone number to each telephone. The CO assigns a list of available numbers for each DID (Direct Inward Dial) line. You can change your DN range to match these numbers, and you use target lines to match each number with a DN.

Figure 2 DID system


Target line mapped to DN (4005)
Target line mapped to DN (4006)
CO DID line i.e. 769 with range of call numbers (4005 to 4020)
Target line mapped to DN (4007)
Target line mapped to DN (4008)

Incoming calls

1 DID trunks are assigned to be auto-answer.

> **Note:** PRI lines are automatically set to auto-answer.

2 All DNs are assigned target lines.

3 A caller dials a system code and a DN. In the example shown above, it might be 769-4006.

4 The call comes into the trunk, which answers and maps the call on the target line assigned to the matching received digits.

5 The DN assigned to that target line rings.

You can assign unanswered or busy telephones to Call Forward to another DN, such as a designated attendant or a voice mail system.

Basic telephony routing

In a basic configuration, simple access codes (for example Line Pool Codes) are used to access the PSTN network.

In a more complex configuration, more advanced destination codes are required to access multiple PSTNs, private network resources, and remote nodes. Access to these resources enables advanced features, such as tandem routing.

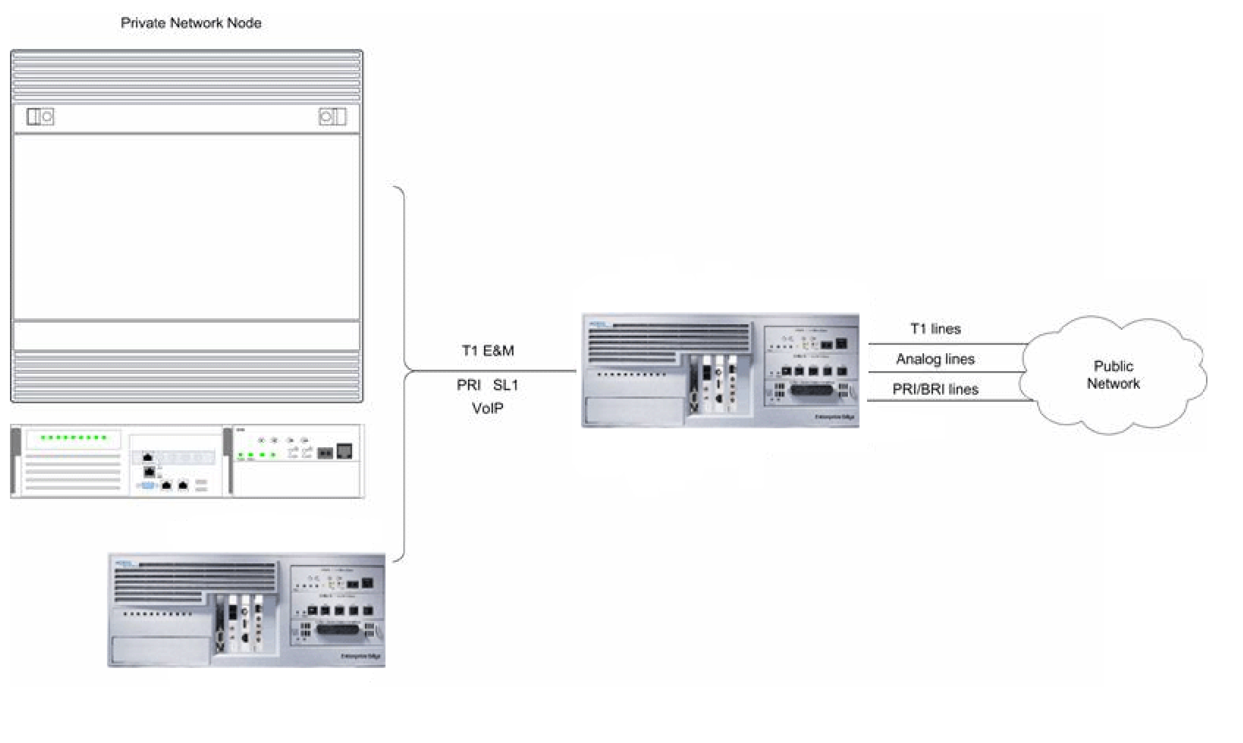
Tandem calling to a remote PSTN

A system connected to a private network that uses dedicated circuits or VoIP circuits can allow a user to dial directly to many other users, on different nodes, using a coordinating dialing plan.

Using a private network saves on toll charges, and local charges, as fewer PSTN accesses are required for internal and external calling. Several nodes located on one site initiate their external local calls to a centralized BCM having a T1 termination to the PSTN. This type of configuration avoids multiple PSTN terminations at other local nodes.

The same tandeming concepts can be applied to inbound calls. DID numbers dialed from the PSTN can be processed and tandem routed out of the centralized system to the localized remote nodes. See other details on Tandem routing "Creating tandem private networks" on page 48.

Figure 3 Tandem dialing through a BCM to/from a private network



In the above example, there are three types of callers.

Each type of caller has a specific method of accessing the other two systems.

Callers using BCM

These callers can:

- call directly to a specific telephone
- select an outgoing line to access a private network
- select an outgoing line to access features that are available on the private network
- select an outgoing central office line to access the public network
- use all of the BCM features

Callers in the public network

These callers use the public lines to:

- call directly to one or more BCM DNs
- call into BCM and select an outgoing TIE line to access a private network
- call into BCM and select an outgoing central office line to access the public network
- call into BCM and use remote features

Callers in the private network node

These callers use the private lines to:

- call directly to one or more BCM DNs
- call into BCM and select an outgoing TIE line to access other nodes in a private network
- call into BCM and select an outgoing central office line to access the public network
- call into BCM and use remote features

System numbering and dialing plans

All systems on a private network must coordinate dialing plans, to ensure that calls get directed to the correct network node. As well, routing becomes more complex, especially if the system is not an end node and must be configured to relay calls to nodes not directly connected to the system. The type of dialing plan supported by the network determines whether each node also requires unique DNs.

Private network parameters

The following provides an overview of the values in the system that affect private networking.

Private networking protocols

The BCM supports the following protocols for private networking:

- PRI: ETSI QSIG, Nortel Voice Networking (MCDN)
- DPNSS (UK only)
- BRI: ETSI QSIG
- T1: E&M
- VoIP trunks (with optional MCDN)

➡ **Note:** Nortel Voice Networking (MCDN) is referred to as SL-1 in Element Manager.

BCM systems can be networked together using T-1, PRI or VoIP trunks. PRI SL-1 lines and VoIP trunks also offer the opportunity to use the MCDN protocol, which provides enhanced trunking features and end-to-end user identification. If a Meridian 1 is part of the Nortel MCDN network, the network can also provide centralized voice mail and auto attendant off the Meridian.

MCDN note: MCDN networking requires all nodes on the network to use a common Universal Dialing Plan (UDP) or a Coordinated Dialing Plan (CDP).

Keycode requirements

Keycodes are required to activate the protocols that are used to create private networking, including:

- VoIP Gateway keycodes
- an MCDN, DPNSS, or Q. Sig keycode, if you want to use a networking protocol between the systems

You must purchase and install these keycodes before you can create a network. Consult with your Nortel distributor to ensure you order the correct keycodes for the type of network you want to create.

Remote access to the network

Authorized users can access TIE lines, central office lines, and BCM features from outside the system. Remote users accessing a private network configured over a large geographical area can avoid toll charges.

➡ **Note:** You cannot program a DISA DN or Auto DN to a VoIP trunk, as they act as auto-answer trunks from one private network to the next. However, you can configure VoIP line pools with remote access packages so that callers can access telephones or the local PSTN on remote nodes on a tandemed network that use VoIP trunks between systems.

Lines used for networking

External (trunk) lines provide the physical connection between BCM and other systems in a private or public network.

The BCM numbers physical lines from 061 to 238. Default numbering depends on the trunk module positioning within the BCM.

VoIP trunks: Although a VoIP gateway does not use physical lines, it is easier to think of them that way. Therefore, in the BCM, lines 001 to 060 are used for VoIP trunk functionality.

BCM networking configurations that use PRI lines, require specific DTM modules.

- DTMs configured for PRI are used for incoming and outgoing calls (two-way DID). Incoming calls are routed directly to a BCM DN that has a properly configured and assigned target line. All outgoing calls made through PRI, are initiated using the destination codes.

- DTMs configured for T1 can have digital lines configured as Groundstart, E&M, Loop, or DID.

Target lines are virtual communication paths between trunks and telephones on the BCM system. They are incoming lines only, and cannot be selected for outgoing calls or networking applications. With target line*s*, you can concentrate incoming calls on fewer trunks. This type of concentration is an advantage of DID lines. BCM target lines allow you to direct each DID number to one or more telephones. VoIP trunks also require target lines to direct incoming traffic. Target lines are numbered 241 to 492.

Telephones can be configured to have an appearance of analog lines or multiple appearances of target lines.

➡ **Note:** PRI B-channels cannot be assigned as line appearances. PRI B-channels, or “trunks”, can only be configured into PRI line pools for inbound routing through target lines with receive digits or outbound routing through destination codes.

Types of private networks

There are several ways you can create private networks. Configuration can be based on such things as cost of trunks, proximity of network nodes, size of the private network, and business requirements for communications.

VoIP-based networking also requires an understanding of IP features such as codecs, jitter buffers, Quality of Service (QoS) function, and silence suppression. Refer to “Silence suppression” on page 695 for more information.

The services provided within networks is based on the type of trunks and the protocols assigned to the trunks. All trunks within the network should be running the same protocols, to provide a technically sound and stable network.

The following links are procedures to set up basic networks to advanced networks, using the support protocols within BCM:

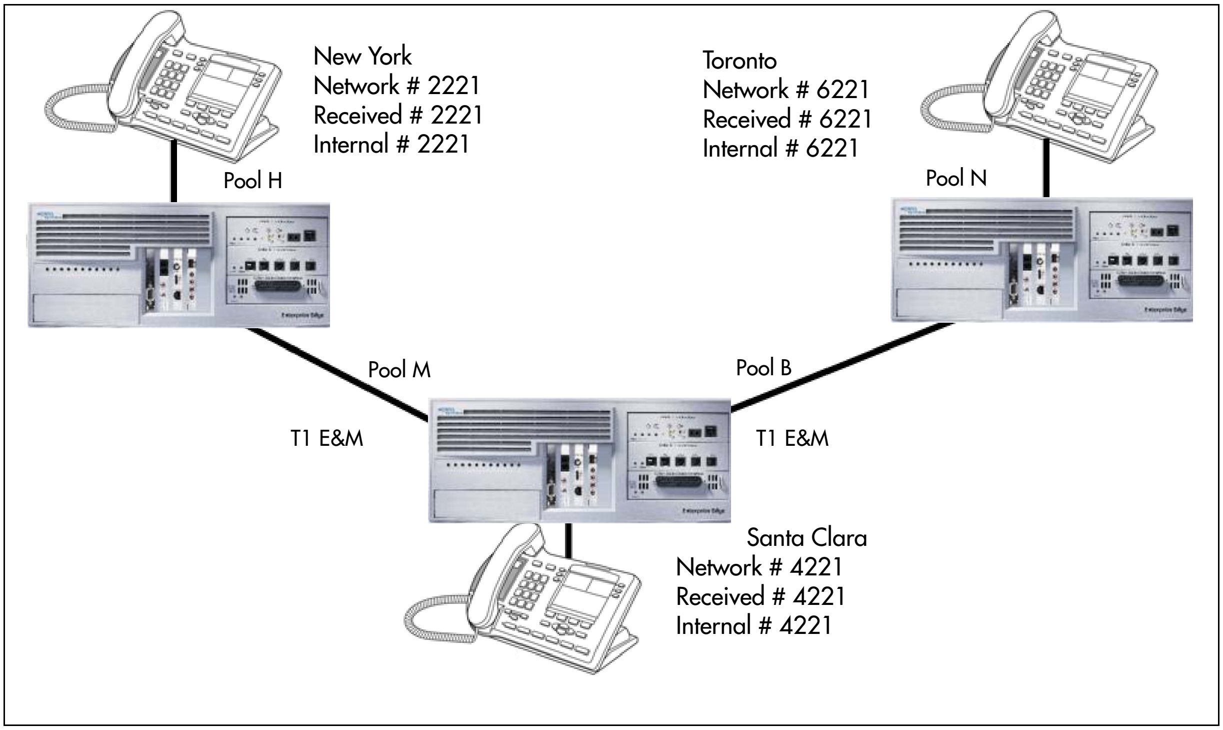
- “Routing-based networks using T1 E&M lines” on page 45
- “PRI networking using Call-by-Call services” on page 47
- “PRI SL-1/Q.Sig/DPNSS and VoIP trunk networking” on page 47

Routing-based networks using T1 E&M lines

By properly planning and programming routing tables and destination codes, an installer can create a dialing plan where T1 E&M lines between BCM systems are available to other systems in the network

Figure 4 shows a network of three BCM systems. Two remote systems connect to a central system.

Figure 4 Dialing plan for T1 E&M routing network



Each system must be running BCM software. Each system must be equipped with target lines and a DTM with at least one T1 E&M line.

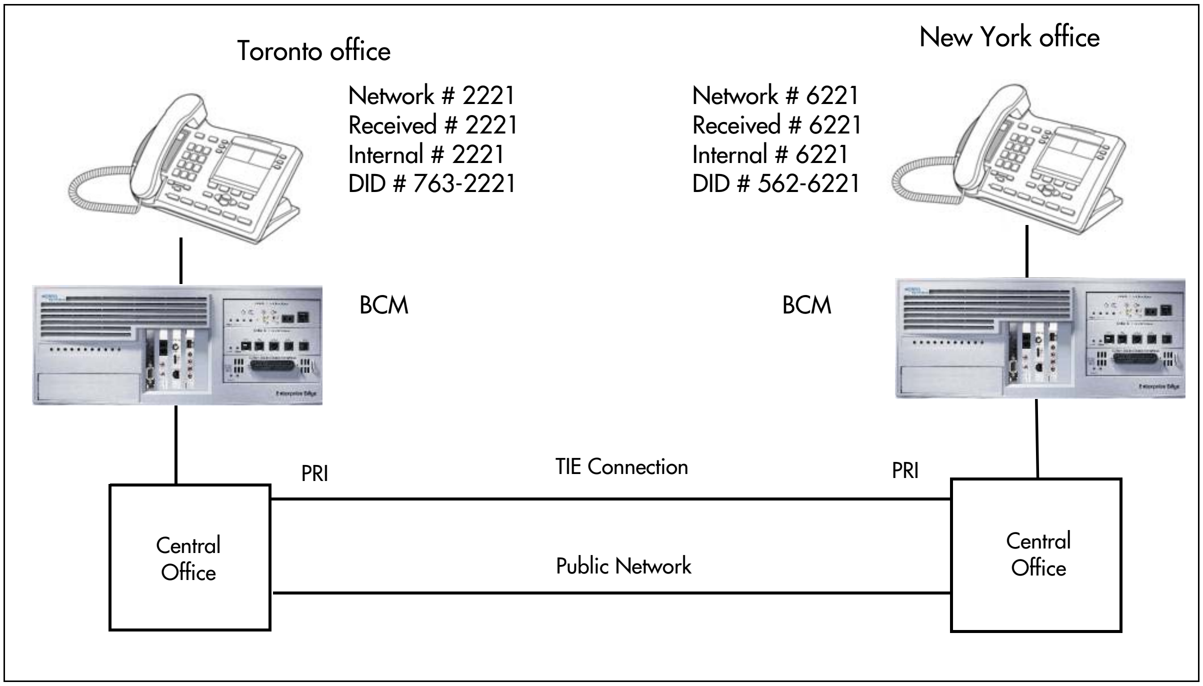
The call appears on the auto answer line on the BCM in Santa Clara as 6-221. Because 6 is programmed as a destination code for Toronto on the Santa Clara system, another call is placed using route 002 from Santa Clara to Toronto. At the Toronto system, the digits 6-221 are interpreted as a target line Private received number. The call now alerts at DN 6221 in Toronto.

➡ **Note:** Network calls that use routes are subject to any restriction filters in effect. If the telephone used to make a network call has an appearance of a line used by the route, the call will move from the intercom button to the Line button. The telephone used to make a network call must have access to the line pool used by the route. Network calls are external calls, even though they are dialed as if they were internal calls. Only the features and capabilities available to external calls can be used.When programming a button to dial a Network number automatically (autodial), network calls must be treated as external numbers, even though they resemble internal telephone numbers. Routes generally define the path between your BCM switch and another switch in your network, not other individual telephones on that switch.

PRI networking using Call-by-Call services

The example shown in Figure 5 highlights the use of PRI Call-by-Call services. It shows two offices of a company, one in New York and one in Toronto. Each office is equipped with a BCM system and a PRI line. Each office has to handle incoming and outgoing calls to the public network. In addition, employees at each office often have to call colleagues in the other office. Refer to "Private networking: PRI Call-by-Call services" on page 377 for more information.

Figure 5 PRI networking using Call-by-Call Services



To reduce long distance costs, and to allow for a coordinated dialing plan between the offices, private lines are used to handle inter-office traffic.

If call-by-call services were *not* used, each BCM system might have to be equipped with the following trunks:

- 12 T1 DID lines needed to handle peak incoming call traffic.
- eight T1 E&M lines needed to handle inter-office calls.
- eight lines needed to handle outgoing public calls

PRI SL-1/Q.Sig/DPNSS and VoIP trunk networking

PRI SL-1 trunks and VoIP trunks can be used to create private networks between BCM systems or between BCM systems and larger call servers such as Meridian 1, Succession 1000/M, DMS100/250 and CSE.

ETSI-QSIG and DPNSS private networking is configured very similarly, although network features may be supported slightly differently due to local line and network requirements.

If the MCDN protocol is added to this type of private network, the network provides additional network management features, as well as allowing centralized voice mail features to be available to all nodes on the network.

The following sections describe the different aspects of SL-1 and MCDN private networking.

- "System dialing plans"
- "Creating tandem private networks"
- "Understanding MCDN network features" on page 51
- "Networking with ETSI QSIG" on page 54
- "Private networking with DPNSS" on page 63

The type of network you require depends on the equipment in the network, and how you want to use the network.

- With MCDN, you can tie a set of BCM systems together with PRI SL-1(MCDN)/ETSI-QSIG, DPNSS or VoIP trunks to create a tandem network. This type of network provides the additional advantage of providing private line access to local PSTNs for all the nodes on the network.

➡ **Note:** A keycode is required to use SL-1(MCDN).

System dialing plans

Both these types of networks require similar setups for dialing plans and routing. Each node must have a way to route external calls to the adjacent node or nodes. To do this, all nodes must have the same Private DN length.

You use routing and a private dialing plan to control calls over the network. Each example in this section describes the routing configurations that are required to support calls over the network.

Depending on the type of dialing plan you choose, each node must also have a unique location or steering code so the calls can be correctly routed through the nodes of the network. MCDN networks also require a Private Network ID, which is supplied by the Meridian network administrator to define how the Meridian system identifies each node.

Creating tandem private networks

You can tie a number of BCM systems together with SL-1 lines. This tandem network provides you with the benefits of end-to-end name display and toll-free calling over the SL-1 private link. Each BCM becomes a node in the network. In this type of network, you must ensure that each BCM system, known as a node of the network, is set up to route calls internally as well as to other nodes on the system. This means each node must have a route to the immediately adjacent node, and the correct codes to distribute the called numbers. Each node must have a unique identification number, which is determined by the type of dialing plan chosen for the network.

As well, you can save costs by having a public network connection to only one or two nodes, and routing external calls from other nodes out through the local PSTN, thus avoiding toll charges for single calls.

VoIP note: You can also use VoIP trunks between some or all of the nodes. The setup is the same, except that you need to create gateway records for each end of the trunk, and routing tables to accommodate the gateway codes.

Routing for tandem networks

In tandem networks, each node needs to know how to route calls that do not terminate locally. To do this, you set up routes for each connecting node by defining destination codes for each route.

If the node is also connected to the public network, the usual routing is required for that connection.

The following tables show the routing tables for Node A and Node C for external and internal terminating calls.

➡ **Note:** The PRI and ETSI QSIG trunks are en bloc dialing lines, so all dialed digits are collected before being dialed out.

Table 1 Node A destination code table, external termination

Route	Absorb length	Destination code (public DNs)
4 (PSTN)	1	91604
3 (Node B)	0	91403762 (Node B)
3 (Node B)	0	91403765 (Node E)
4 (PSTN)	1	9140376* (not internal network)
4 (PSTN)	1	914037* (not internal network)
4 (PSTN)	1	91403* (not internal network)
4 (PSTN)	1	9* (not internal network)
* This wild card represents a single digit.		

Table 2 Node A destination code table, internal termination

Route	Absorb length	Destination code (private DNs)
3 (Node B)	0	392 (Node B)
3 (Node B)	0	395 (Node E)
5 (Node C)	0	393 (Node C)
5 (Node C)	0	394 (Node D)
5 (Node C)	0	396 (Node F)

Table 3 Node C destination code table, external termination

Route	Absorb length	Destination code (Public DNs)
3 (Node B)	0	91613764 (Node D)
3 (Node B)	0	91613766 (Node F)
4 (PSTN)	1	9161376* (not internal network)
4 (PSTN)	1	916137* (not internal network)
4 (PSTN)	1	91613* (not internal network)
4 (PSTN)	1	9161* (not internal network)
4 (PSTN)	1	916* (not internal network)
4 (PSTN)	1	91* (not internal network)
4 (PSTN)	1	9 (not internal network)
* This wild card represents a single digit.		

Table 4 Node C destination code table, internal termination

Route	Absorb length	Destination code (Private DNs)
3 (Node D)	0	394 (Node D)
3 (Node D)	0	396 (Node F)
5 (Node A)	0	391 (Node A)
5 (Node A)	0	392 (Node B)
5 (Node A)	0	395 (Node E)

Understanding MCDN network features

When you connect your BCM systems through PRI-SL-1/ETSI QSIG/DPNSS or VoIP trunks, and activate the MCDN protocol, your network provides a number of network call features. You can use this protocol to network other BCM systems, such as the tandem system shown in the previous section, Norstar systems, Meridian 1 systems, Succession systems, DMS 100 systems or CSE systems.

Table 5 lists the MCDN features that are provided by all SL-1/VoIP networks where MCDN is active. The features affect call redirection and trunking functions.

Table 5 MCDN network features

Centralized messaging	"Network Call Redirection Information" (NCRI)
Centralize trunking	"ISDN Call Connection Limitation" on page 52 (ICCL) "Trunk Route Optimization" on page 53 (TRO) "Trunk Anti-tromboning" on page 53 (TAT)

Network Call Redirection Information

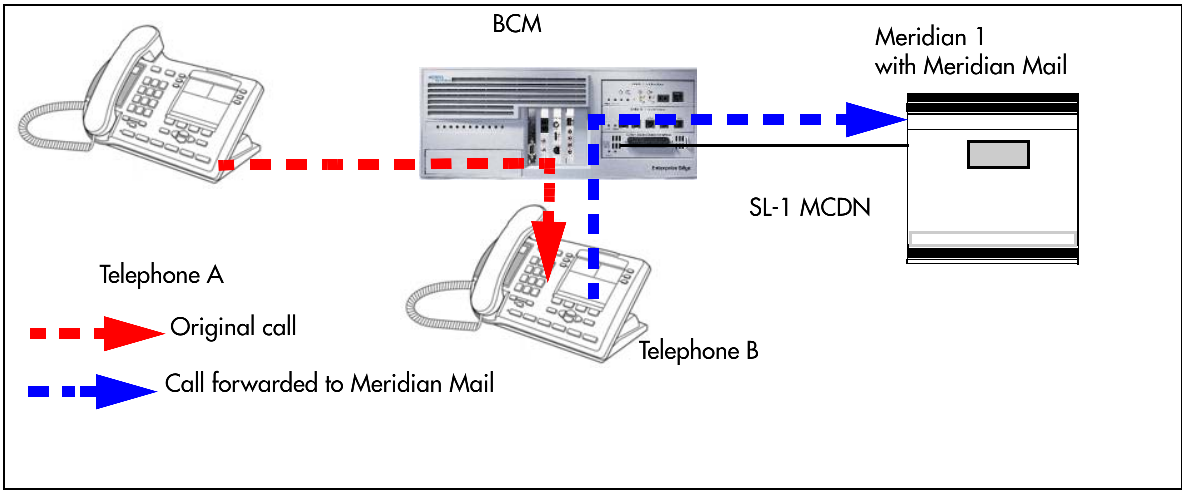
NCRI (Network Call Redirection Information) builds on the following BCM features:

- External Call Forward
- Call Transfer
- Call Forward

NCRI adds the ability to redirect a call across an MCDN network using Call Forward (All Calls, No Answer, Busy) and Call Transfer features. The call destination also receives the necessary redirection information. This feature allows the system to automatically redirect calls from within a BCM system to the mail system, such as Meridian Mail, which resides outside the BCM system on the Meridian 1.

Figure 6 shows an example of this situation, where user A calls user B on the same BCM. If user B is busy or not answering, the call automatically gets transferred to a Meridian Mail number (user C) across an MCDN link between the BCM system and the Meridian 1 system where the mailboxes are set up.

Figure 6 Network call redirection path



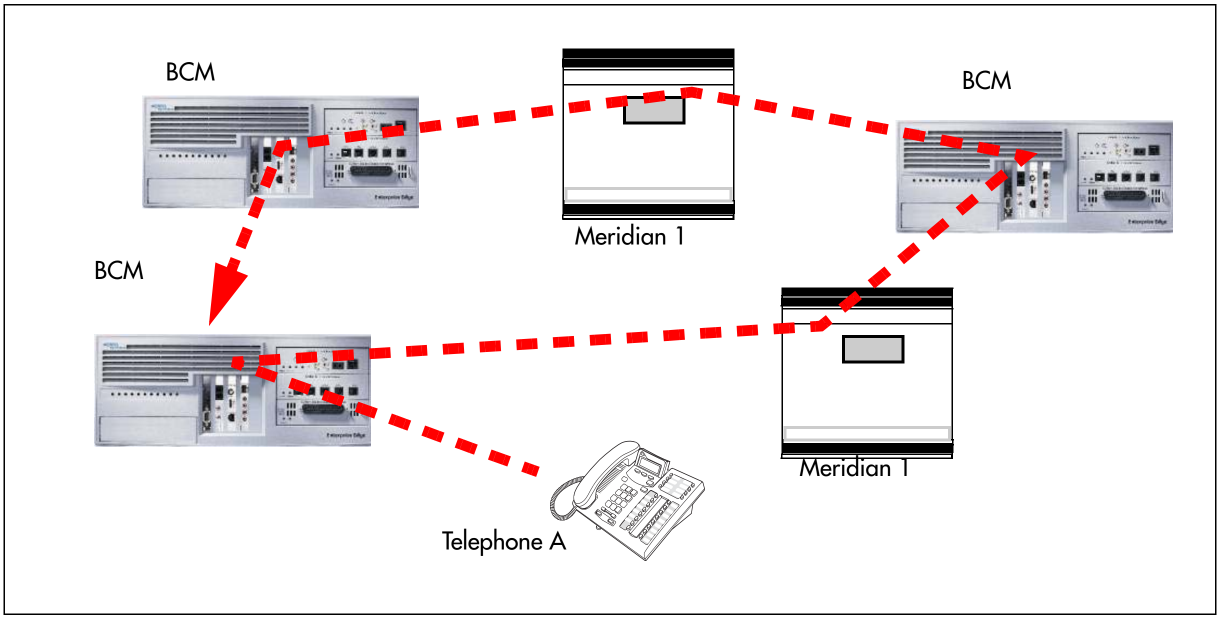
ISDN Call Connection Limitation

The ICCL (ISDN Call Connection Limitation) feature piggybacks on the call initiation request and acts as a check at transit PBX points to prevent misconfigured routes or calls with errors from blocking channels. Also refer to "ISDN overview" on page 701.

This feature adds a transit/tandem counter to a call setup message. This counter is compared at each transit PBX with a value programmed into the transit PBX, in a range from 0 to 31. If the call setup counter is higher than the PBX value, the call will be blocked at the PBX system and cleared back to the network. This prevents calls from creating loops that tie up lines.

Figure 7 demonstrates how a call might loop through a network if the system is not set up with ICCL.

Figure 7 Call loop on system without ICCL



Trunk Route Optimization

Trunk Route Optimization (TRO) finds the most direct route through the network to send a call between nodes. This function occurs during the initial alerting phase of a call.

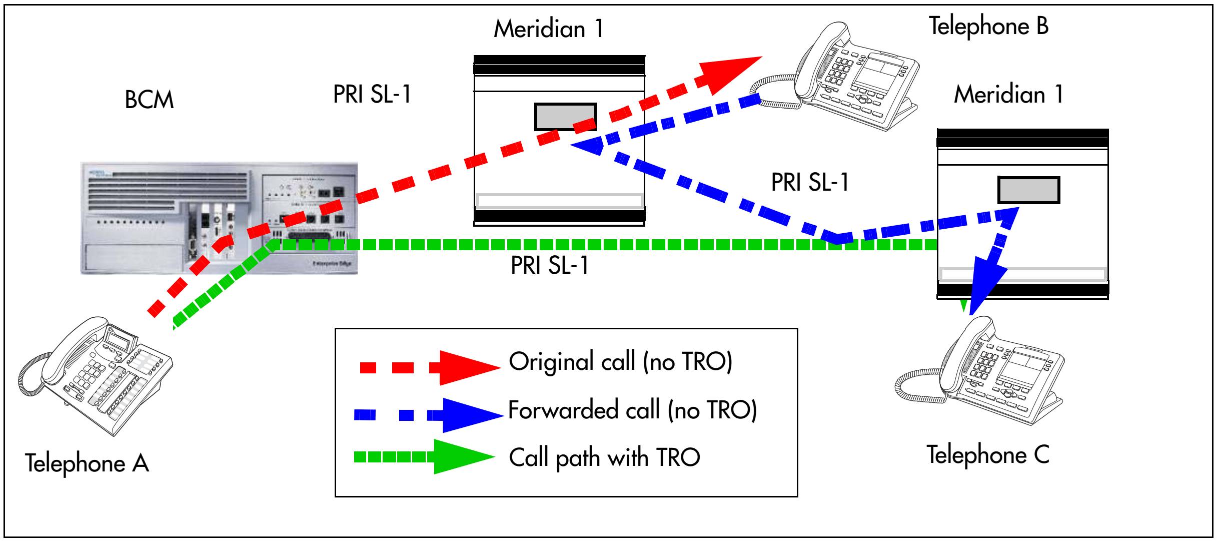
BCM configurations:

- Under **Configuration > Dialing Plan > Private Network**, select the check box beside **TRO**.
- Configure call routing for all optimal routes.
- Configure call forward (All Calls, No Answer, Busy) or Selective Line Redirection to use the optimal routes.

This feature avoids the following situation: A call originating from a BCM system may be networked to a Meridian system, which, in turn, is networked to another Meridian system, which is the destination for the call. If the call routes through the first Meridian (M1) to reach the second Meridian (M2), two trunks are required for the call. An optimal choice is a straight connection to M2. This finds these connections and overrides the less-efficient setup.

Figure 8 shows two call paths. The first route, through the Meridian, demonstrates how a call might route if TRO is not active. The second route, that bypasses the Meridian, demonstrates how TRO selects the optimum routing for a call.

Figure 8 Call paths with and without TRO



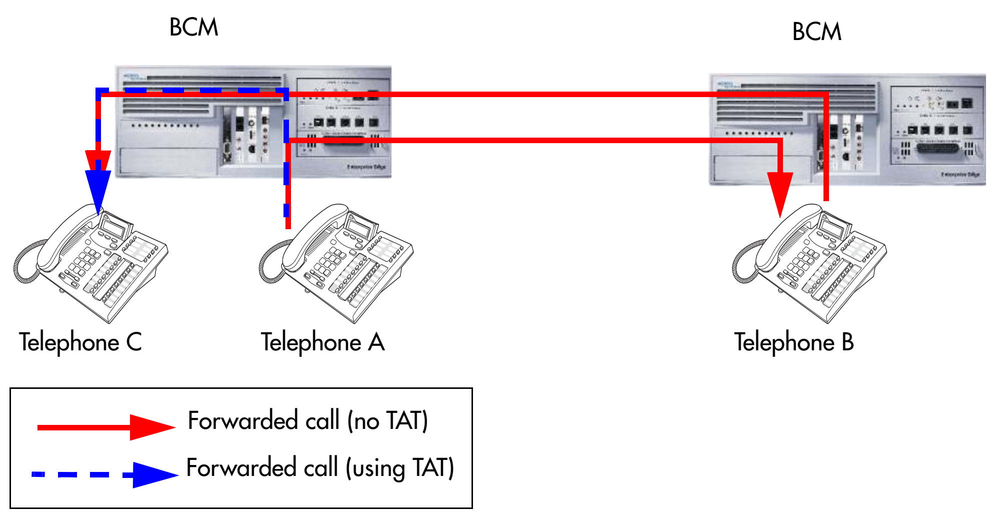
Trunk Anti-tromboning

Trunk Anti-Tromboning (TAT) is a call-reroute feature that works to find better routes during a transfer of an active call. This feature acts to prevent unnecessary tandeming and tromboning of trunks.

Note: This feature is not applicable for alerting calls.

Figure 9 shows how TAT reduces the line requirements. The solid line shows Telephone A calling Telephone B and being transferred over an additional PRI line to Telephone C. With TAT active, the same call is transferred to Telephone C over the same PRI line.

Figure 9 Call paths with and without TAT



Networking with ETSI QSIG

(ETSI QSIG applies only to international systems equipped with a DTM or BRIM. It is not supported on VoIP)

ETSI QSIG is the European standard signaling protocol for multi-vendor peer-to-peer communications between PBX systems and/or central offices.

Other information in this section: “ETSI Euro network services” on page 55.

Figure 10 illustrates an ETSI QSIG network. Note that this is exactly the same setup as that shown in the MCDN section for North America. The hardware programming for ETSI QSIG is described in Table 6. All other configurations are the same as those shown in the MCDN section for North America.

Figure 10 ETSI QSIG networking

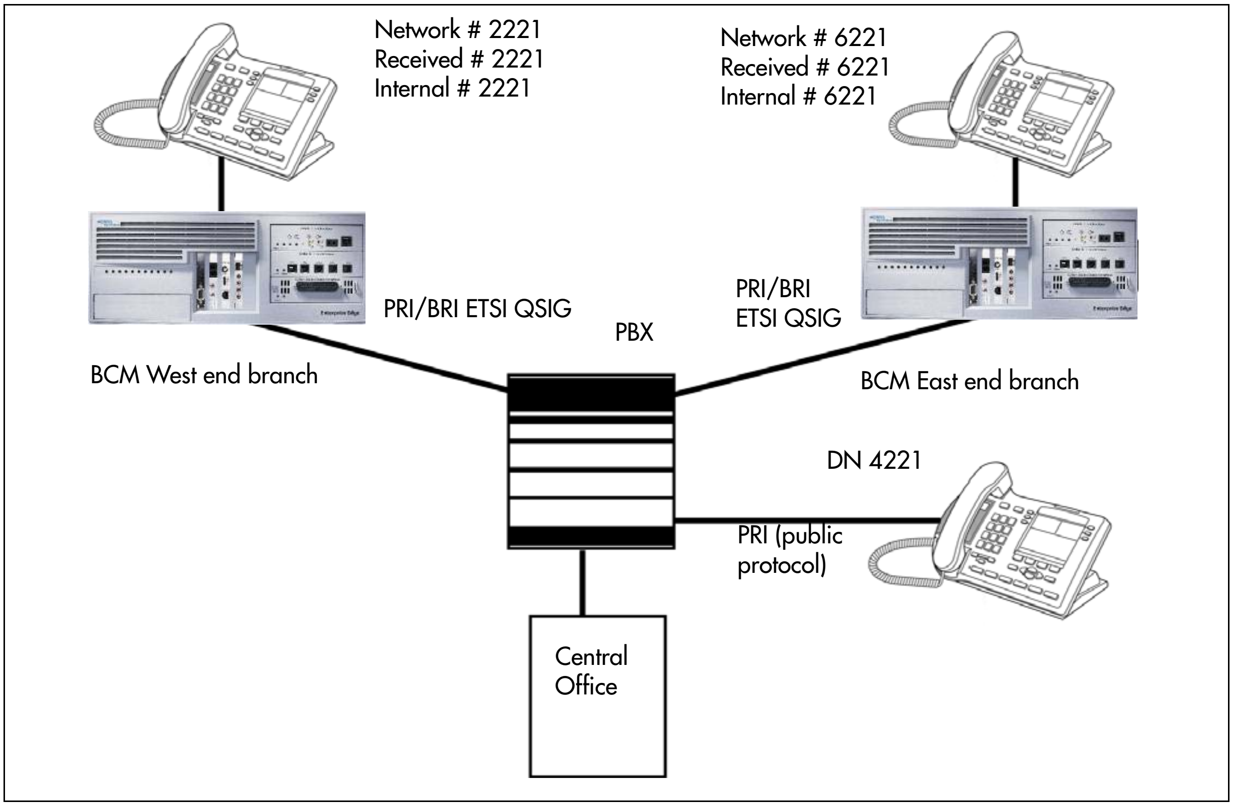


Table 6 lists the settings for some of the hardware parameters for ETSI QSIG networking example shown in Figure 10.

Table 6 Hardware programming for branch offices

West End office:		
Hardware programming	DTM/BRIM	PRI/BRI
	Protocol	ETSI QSIG
	BchanSeq	Ascend (PRI only)
	ClockSrc	Primary

East End office:		
Hardware programming	DTM/BRIM	PRI/BRI
	Protocol	ETSI QSIG
	BchanSeq	Ascend (PRI only)
	ClockSrc	Primary

ETSI Euro network services

If your system has ETSI ISDN BRI/PRI lines, you can activate the malicious call identification (MCID) and Network Diversion features. Advice of charge-end call (AOCE) is active if your service provider has activated that service on the line.

When the features are activated, users can:

- display a call charge
- redirect calls over the ETSI ISDN BRI/PRI line to the outside network
- tag malicious calls

Advice of Charge-End of Call (AOCE) — AOCE is a supplementary service available from your service provider on ETSI ISDN BRI/PRI links. This feature allows the BCM user to view the charges for an outgoing call once the call completes. This information is also reported to the Call Detail Reporting Application. The information can be provided in currency or charging units, depending on how the feature is set up by your service provider.

To invoke the feature, the user presses **FEATURE 818**.

DPNSS 1 services

The Digital Private Network Signaling System (DPNSS 1) is a networking protocol enhancement that extends the private networking capabilities of existing BCM systems. It is designed to offer greater centralized functionality for operators, giving them access to BCM features over multiple combined networks.

Note: The DPNSS feature is dependent on which region is loaded on your system at startup and requires that a software keycode was entered to enable the feature. The feature also requires a DTM-based connection.

Refer to the following topics:

- "DPNSS 1 capabilities"
- "DPNSS 1 features" on page 57
- "Private networking with DPNSS" on page 63

DPNSS 1 allows a BCM local node, acting as a terminating node, to communicate with other PBXs over the network. For example, corporate offices separated geographically can be linked over DPNSS 1 to other BCM nodes, bypassing the restrictions of the PSTNs to which they may be connected. This allows connected BCM nodes to function like a private network, with all features of BCM accessible.

Note: BCM DPNSS 1 works as a terminating node only. BCM to BCM DPNSS is not supported.

DPNSS 1 features can be used on any BCM telephone. On most BCM telephones, you must use specific keys and/or enter a number code to access the features.

DPNSS 1 capabilities

A single BCM node, acting as a terminating node on the network, supports the following capabilities over DPNSS 1 lines:

- Direct Dial Inward (DDI) for incoming calls.

- Originating Line Identification (OLI) for incoming and outgoing calls:
  - — For incoming calls, the Calling Line Identification (CLI/CLID) information is displayed to the user on telephones with line display. This must be configured in programming.
  - — For outgoing calls, the directory number of the originating party is sent out as OLI.
- Terminal Line Identification (TLI) for incoming and outgoing calls. Referred to as Called Line Identification.
- Selective Line Redirect (SLR) and External Call Forward (ECF) implemented on calls between DPNSS 1, and BRI/PRI, DASS2, and analog lines.
- These remote access features are supported on DPNSS: DDI, line pool access code, destination codes, and remote page feature codes.

Note: Keycodes are required to enable DPNSS 1.

DPNSS to Embark connections

DPNSS lines connected to an Embark switch perform call redirection/diversion using the Call Forward feature to create a tandem link back to the switch. Since this is different from other switches, you must select the type of switch DPNSS will be connecting to when you do module programming.

Before you program Call Forwarding, ensure that:

- Both real channels and virtual channels are provisioned.
- Destination or line pool codes are programmed for the DPNSS to Embark link.

Also, during programming for Call Forward No Answer and Call Forward on Busy, when you enter the **Forward to:** digits, the system performs a validation check with the switch on the number. (**Configuration > Telephony > Sets > Active Sets > Line Access > Properties tab**)

DPNSS 1 features

The following features are available and can be programmed over DPNSS lines:

- “Three party service” on page 58
- Diversion (“Using the diversion feature” on page 58)
- Redirection (“Using the Redirection feature” on page 60)
- “Executive intrusion” on page 60
- “Call Offer” on page 61
- “Route optimization” on page 62
- “Loop avoidance” on page 63
- Message Waiting Indication

The following parameters can be configured for DPNNS 1 lines:

- Line type
- Prime set

- CLID set
- Auto privacy
- Answer mode
- Auxiliary ringer
- Full autohold

Some features are transparent to the user, but must be programmed to be activated. Others are available for end-user programming at the telephone. Details about these features are given below.

Three party service

Three Party Service is a DPNSS 1 feature for BCM that is similar to the BCM Conference feature.

The Three Party Service allows a user, usually an operator, to establish a three-party conference by calling two other parties from one telephone. Once the connection is made, the controlling party can hang up, leaving the other two connected. The controlling party can even put one party on hold, and talk to the other party.

Note: BCM does not support Hold over the DPNSS link itself. This means that the conferenced party on the distant end of the network cannot place a Three Party Service call on Hold.

This feature is designed to allow operators to assist in the connection of calls from one main location.

Making a conference call

To initiate or disconnect from a conference call on a BCM system over DPNSS 1, use the procedure described in the *BCM 4.0 Device Configuration Guide* (N0060600).

Note: Three Party Service is supported on model 7000 telephones, but in a receive-only fashion. These telephone types cannot initiate Three Party Service. For more information about these telephone types, see the *BCM 4.0 Telephony Device Installation Guide* (N0060609).

(model 7000 phones, supported in Europe only.)

Using the diversion feature

Diversion is a DPNSS 1 feature for BCM that allows users to forward their calls to a third party on the DPNSS 1 network. This feature is similar to Call Forward on BCM, but takes advantage of the broader capabilities of DPNSS.

There are five variations of Diversion: Call Diversion Immediate, Call Diversion On Busy, Call Diversion On No Reply, Bypass Call Diversion, and Follow-me Diversion. These variations are described below:

- Diversion Immediate diverts all calls to an alternate telephone. This function is programmed by the user at their telephone.

- Diversion On Busy diverts all calls to an alternate telephone when a telephone is busy. This feature is programmed in the Element Manager.
- Diversion On No Reply diverts calls that go unanswered after a specified amount of time. This feature is programmed in the Element Manager.
- Bypass Call Diversion overrides all call forward features active on a telephone over a DPNSS line. An incoming call to the telephone will not be forwarded; instead, the telephone will continue to ring as if call forward were not active. This feature is used to force a call to be answered at that location. Bypass Call Diversion is a receive-only feature on BCM, and cannot be used from a BCM telephone.
- Follow-me Diversion is also a receive-only feature. It allows the call forwarded destination to remotely change the BCM call forwarding programming (Call Forward All Calls [CFAC] feature) to a different telephone.

Note: BCM CFAC must be active and the destination set/PBX system must support the feature.

For example, user A forwards all calls to telephone B, a temporary office. Later, user A moves on to location C. The user does not have to be at telephone A to forward calls to location C. Using telephone B and Follow-me Diversion, the user can forward calls from A to location C.

Follow-me diversion can be cancelled from the forwarded location.

- Diversion on Busy and Diversion on No Reply cannot be cancelled from the forwarded telephone. These are programmable only by an installer and not by the user.
- If multiple telephones are programmed to take a call, the first telephone to respond will act. All other telephones responding are ignored. Therefore, if the first telephone to respond has Diversion enabled, this feature will be invoked.

Restrictions by telephone type

- all variations supported on BCM digital and IP telephones
- ATA2/ASM8+—all variations supported on an ATA
- ISDN—all variations supported on ISDN telephones, except Diversion on Busy and CFWD Busy

Setting Diversion

You set Diversion for DPNSS in the same way as Call Forward. You will need to enter the end DN when prompted. You may also need to include the DPNSS 1 routing number.

Using the Redirection feature

Redirection is a DPNSS 1 feature similar to BCM Transfer Callback. Redirection lets a call awaiting connection, or reconnection, be redirected by the originating party to an alternate destination after a time-out period. Failed calls can also be redirected. Priority calls are not redirected.

Note: The address to redirect depends on the history of the call. Calls that have been transferred are redirected to the party that transferred them. In all other cases, the address to redirect is the one registered at the PBX system originating the redirection.

Note: BCM does not support the redirection of BCM originated calls, even over DPNSS 1.

The Diversion on No Reply feature takes precedence over Redirection.

Restrictions by telephone type

- For telephones with a single line display, the number key (#) acts as `MORE` and the star key (*) acts as `VIEW`
- ISDN—all variations supported on ISDN telephones

Setting redirection

The timer used for the network Callback feature is also used for redirection.

Executive intrusion

Executive Intrusion (EI) is a DPNSS 1 feature that allows an operator, or other calling party, to intrude on a line when it is busy. An example of the use of this feature is to make an important announcement when the recipient is on another call.

EI is similar in functionality to BCM Priority Call, but it is a receive-only feature on BCM telephones. EI cannot be initiated from a BCM telephone. The person using this feature must be on another PBX system on the DPNSS 1 network.

When EI is used to intrude on a call in progress, a three-way connection is established between the originating party and the two parties on the call. The result is very much like a conference call. When one of the three parties clears the line, the other two remain connected, and EI is terminated.

Restrictions by telephone type

- ATA2/ASM8+—supported
- ISDN—not supported

The telephone receiving the intrusion displays `Intrusion Call`. A warning indication tone will sound after intrusion has taken place, and the standard conference call tone will sound every 20 seconds.

Intrusion levels

Whether or not a telephone will accept or reject an Executive Intrusion request depends on the level of intrusion protection programmed. Each telephone (DN) has an Intrusion Capability Level (ICL) and four Intrusion Protection Levels (IPL).

When the ICL of the intruding telephone is higher than the IPLs of *both* telephones on the active call, EI occurs. Nortel recommends setting the IPLs of most BCM telephones to the default of None, Low, or Medium.

Intrusion levels are described as follows:

- ICL: determines the ability of the attendant to intrude. As long as the ICL is higher than the IPL of the wanted party, EI is allowed. Since EI is a receive-only feature, the ICL cannot be set on BCM.
- IPL: determines the ability of the attendant to refuse intrusion. If the IPL is lower than the ICL of the originating party, EI is allowed. For general purposes, Nortel recommends setting the IPL to None, Low, or Medium, unless intrusion is not wanted.

Call Offer

Call Offer over DPNSS 1 allows a calling party to indicate to the wanted party that there is an incoming call available, even though there is no answer button available to present the call on the telephone. The intended recipient can ignore, accept, or decline the offered call. Call Offer is useful in increasing the call-coverage capability of a BCM system, and helps to lift the network processing load. It is a receive-only capability on BCM: incoming calls would be initiated at another PBX system on the DPNSS 1 network.

An example of Call Offer in use is an operator or attendant who has a number of calls coming in at once. The operator can call offer one call and move to the next without waiting for the first call to be answered.

Call Offer Displays

When a Call Offer is made by the originating exchange, the target telephone displays a message, and a tone is heard. When an offered call arrives on telephones with line display, the user sees `XX...X wtng` if the calling party ID is available and CLID is enabled. If CLID is not available or CLID is disabled, `Line XXX waiting` appears (the line name associated with the call). If there are more than 11 digits in the incoming number, only the last 10 will display.

If Call Queuing is programmed for the system, the display shows `Release Line XXX`.

This is the line name of the highest-priority queued call if it is an offered call.

Restrictions by telephone type

- model 7000 telephone — associated LED or LCD flashes, and a tone is heard (model 7000 phones, supported in Europe only.)
- ATA2/ASM8+—Call Offer is supported as a Camp On feature, and a tone is heard
- ISDN—not supported

Note the following general conditions and restrictions:

- DND on busy must be selected (**DN > Capabilities and Preferences > Capabilities** tab) for a telephone to accept Call Offer.
- If CF on busy is programmed for the telephone, Call Offer is not accepted.
- The target line for the telephone must be set to: If **busy: busy tone**, which is the default.
- Call Offer does not work if sent over Manual answer lines. It is recommended that the lines be left at the default: **Auto**.

User actions

The party receiving a Call Offer has three choices:

- Ignore it. After a programmed time interval, the Offer request is removed.
- Reject it. If the user activates Do Not Disturb on Busy (DND) when the Call Offer request is made, the request is removed from the telephone. The calling party is informed of the rejection.

Note: A call cannot be offered to a telephone with DND active. The line indicator for external incoming calls still flashes.

- Accept it. The Offer is accepted by releasing the active call.

Note: Forward on Busy takes priority over DND on Busy. Call Offer cannot be accepted by putting an active call on hold.

Route optimization

Route Optimization is a DPNSS 1 feature for BCM that allows calls to follow the optimum route between two end PBXs. This allows efficient use of network resources.

Route Optimization is initiated by the system and is transparent to the user. However, the user may see a call switch from an appearance on the telephone to another appearance key or from an intercom button to the appearance key or vice versa. This occurs when BCM receives a Route Optimization request and initiates a new call to follow the optimal route.

If a telephone is active on a private line call, the Route Optimization call being established may go on a public line. This will cause a loss of privacy on that line.

Data calls are rejected by Route Optimization in order to ensure the data transmission is not affected.

Certain situations result in Route Optimization not taking place. For example, calls that are using Hold, Parking or Camp features do not undergo Route Optimization, and if a Route Optimization call undergoes Diversion, the Route Optimization is dropped.

Setting Route Optimization

There is no system programming required for the feature when BCM is working as a terminating PBX system. However, BCM must have a private access code programmed that maps to a valid destination code or line pool code on DPNSS lines. Further, **Allow Redirect** must be selected.

Loop avoidance

Errors in the configuration of a network may make it possible for a call to be misrouted, and arrive at a PBX system through which it has already passed. This would continue, causing a loop which would eventually use up all of the available channels. The Loop Avoidance service permits counting of DPNSS 1 transit PBXs and rejecting a call when the count exceeds a predetermined limit.

Private networking with DPNSS

(International only)

DPNSS supports the Universal Dialing Plan (UDP), an international standard for sending and receiving private numbers over networks. The UDP requires that a dialing number include the following:

- a Private Access Code, programmed into the system as part of the destination code table to prevent conflicts with the internal numbering system. (**Access Codes**)
- a Home Location Code (HLC) assigned to each PBX system, and configured as part of the destination code (a maximum of seven digits). For each HLC, a destination code must be programmed in the system. (**Configuration > Telephony > Dialing Plan > Private Networking**)
- a Directory Number (DN) assigned to each extension as a line appearance. The DN appears as the last string segment in a dialed number. In the number 244-1111, 1111 is the DN.

A typical Private Number, using a private access code and dialed from another site on the network, appears below.

Private Access Code	+ Home Location Code	+ Directory Number	= Calling Party Number
6	+ 848	+ 2222	= 6-848-2222

In this networking example, a private network is formed when several systems are connected through a Meridian 1 and a terminating BCM system. Each site has its own HLC and a range of DNs. Figure 11 illustrates this example.

Table 7 shows examples of the construction of numbers used when dialing within the example network. Note that 6 is the Private Access code.

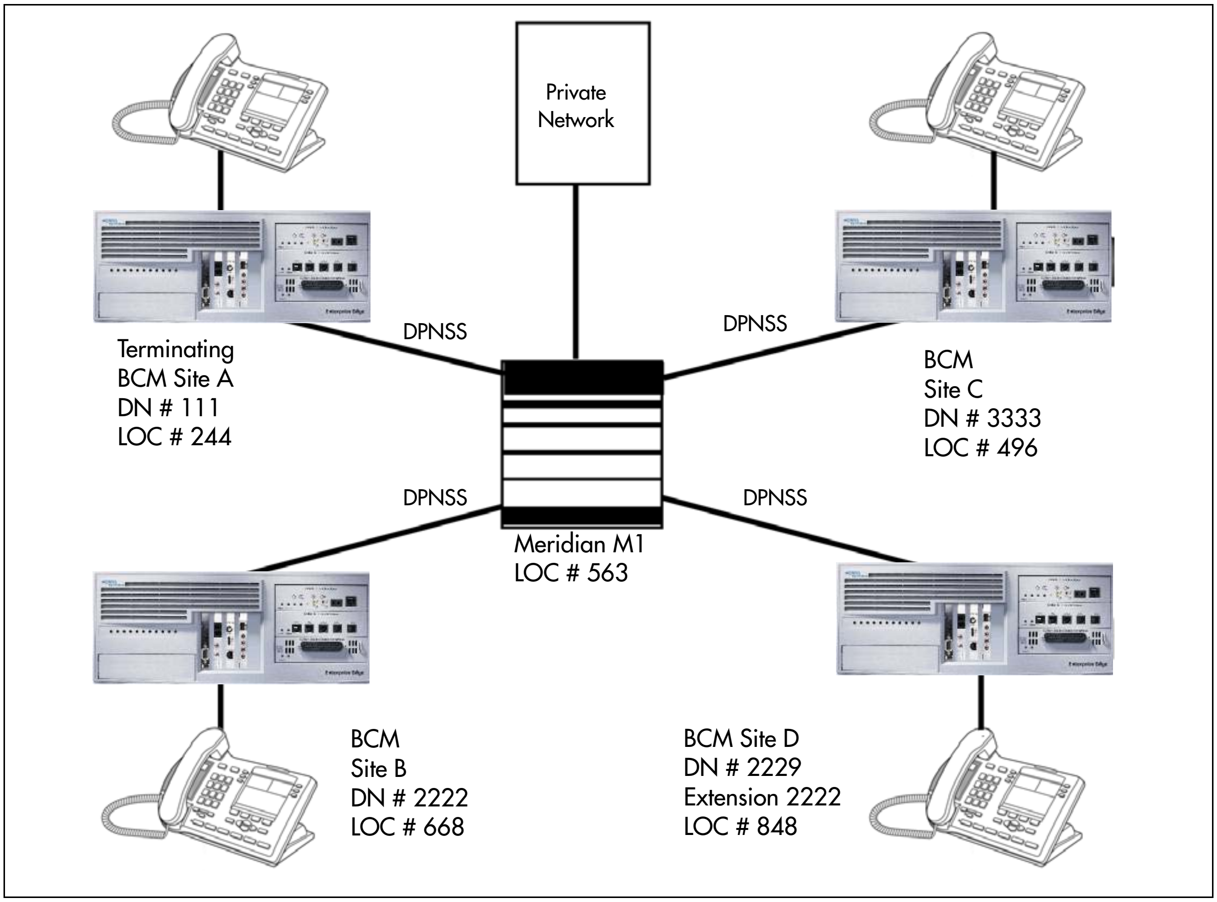
Table 7 Calling numbers required for DPNSS network example (Sheet 1 of 2)

Calling Site	LOC/HLC	Calling Party Number	Called Site	Dialing String	Called Party Number
Site A	244	244 1111	Site B	6 668 2222	668 2222
Site B	668	668 2222	Site D	6 848 2222	848 2222

Table 7 Calling numbers required for DPNSS network example (Sheet 2 of 2)

Calling Site	LOC/HLC	Calling Party Number	Called Site	Dialing String	Called Party Number
Site D	848	2222	Site D	2229	2229
Site C	496	496 3333	Public DN	9 563 3245	563 3245

Figure 11 DPNSS networking


Private Network
DPNSS
DPNSS
Terminating BCM Site A DN # 111 LOC # 244
BCM Site C DN # 3333 LOC # 496
DPNSS
DPNSS
Meridian M1 LOC # 563
BCM Site B DN # 2222 LOC # 668
BCM Site D DN # 2229 Extension 2222 LOC # 848

Calls are dialed and identified to the system as follows:

- To reach a telephone inside the Private Network, at the BCM site, the user dials the DN of choice.
- To reach a telephone inside the Private Network, from another site, the user dials HLC + DN.
- To reach a telephone outside the Private Network, the user dials an Access Code + HLC + DN.

  Each node has its own destination (dest) code, which includes the appropriate access and HLC codes to route the call appropriately.

Chapter 3

Telephony programming: Configuring call traffic

Telephony call traffic has a number of configuration requirements. Some configuration is common to both incoming and outgoing traffic. Other settings are specific to the call direction.

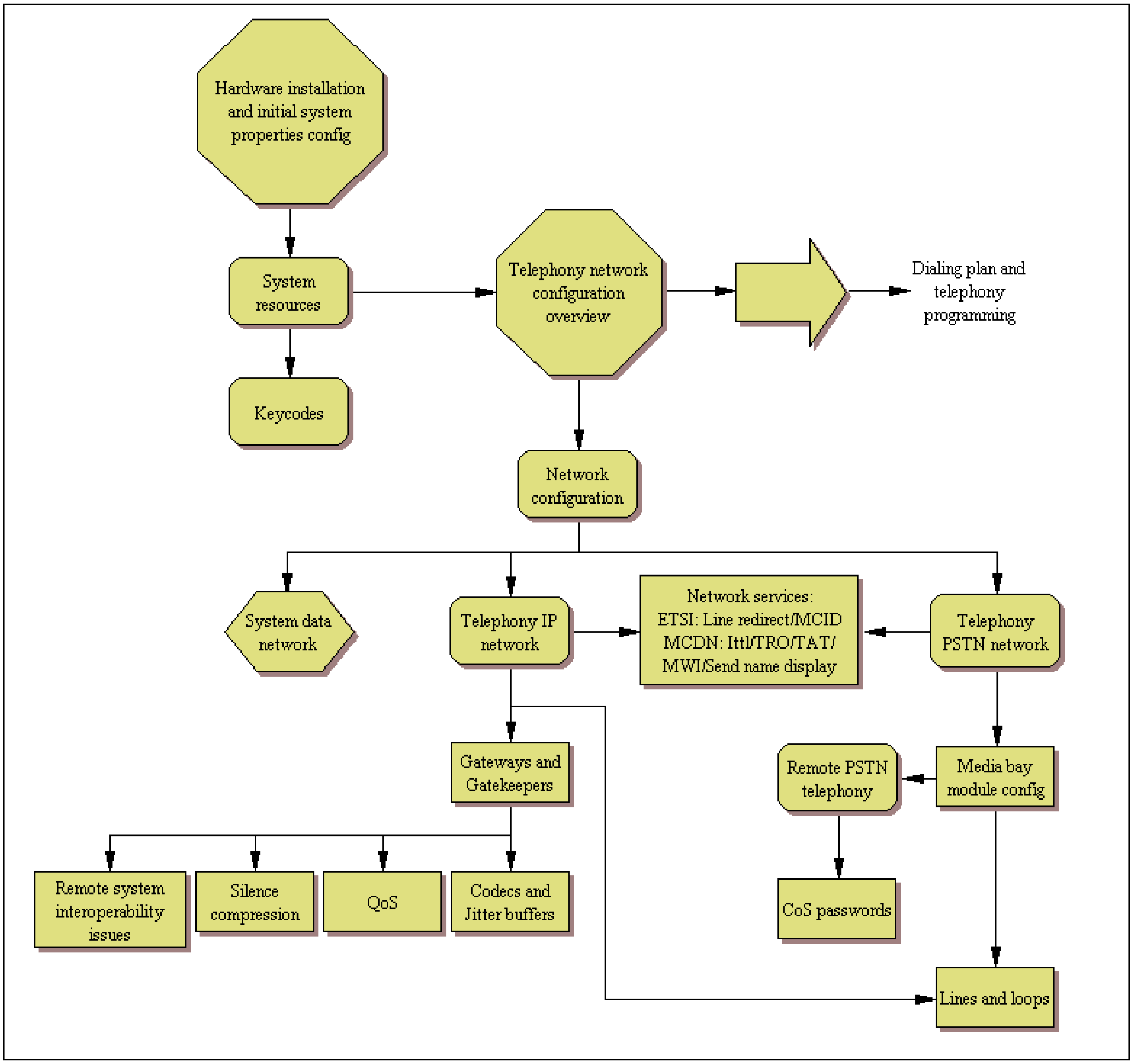
In the case of private networking, call configuration becomes more complex, as remote systems send calls over the private network to other nodes or to your system PSTN network and your local PSTN handles calls directed to remote nodes through your system.

Line programming and number planning both play critical roles in controlling call traffic for your system.

Refer to the following topics:

- "Incoming calls" on page 68
- "Outgoing calls" on page 72

Figure 12 Telephony system and device programming


Hardware installation and initial system properties config
System resources
Keycodes
Telephony network configuration overview
Dialing plan and telephony programming
Network configuration
System data network
Telephony IP network
Network services: ETSI: Line redirect/MCID MCDN: Ittl/TRO/TAT/ MWI/Send name display
Telephony PSTN network
Gateways and Gatekeepers
Remote PSTN telephony
Media bay module config
Remote system interoperability issues
Silence compression
QoS
Codecs and Jitter buffers
CoS passwords
Lines and loops

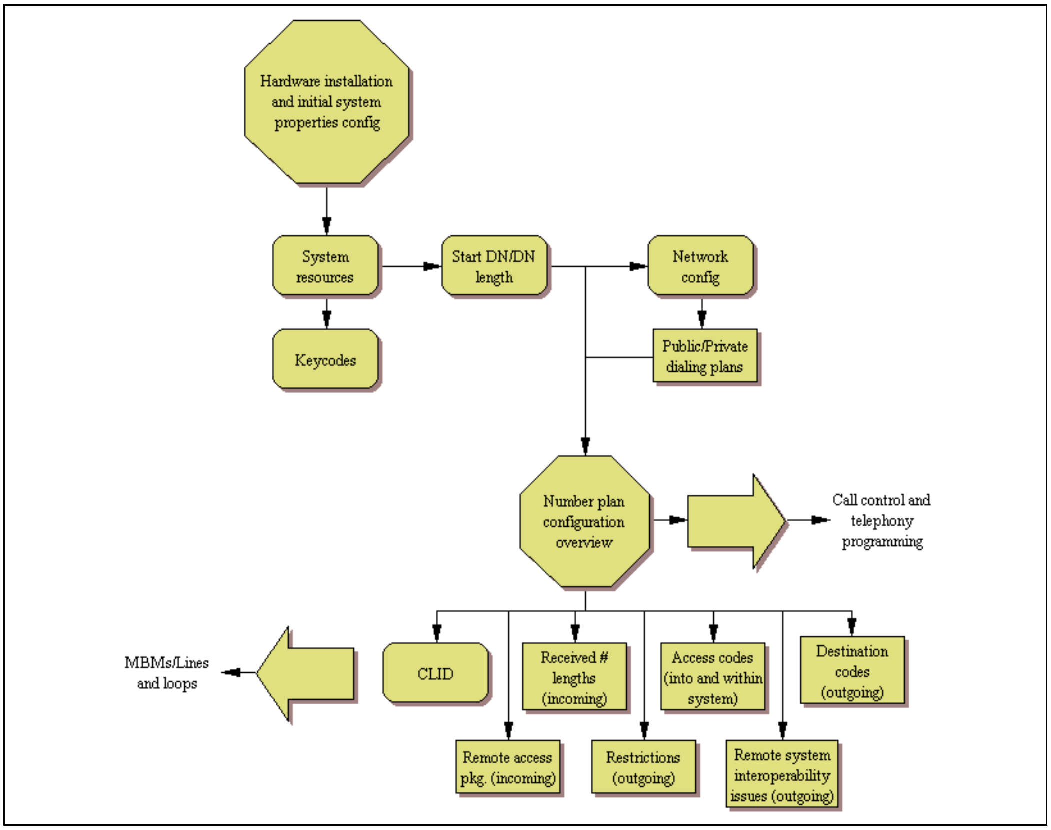
Although many of the tasks involved in programming both areas can be performed in any order, work flow falls generally in the following order:

- Module configuration/VoIP trunk gateways
  - — "Configuring telephony resources" on page 119
  - — "Managing modules" on page 117
  - — "Configuring the trunk module to line type" on page 107
  - — "Configuring VoIP trunk gateways" on page 409
  - — "VoIP interoperability: Gatekeeper configuration" on page 417
  - — "Setting up VoIP trunks for fallback" on page 423
- Line configuration/target line configuration
  - — "BRI ISDN: BRI loop properties" on page 227

- — "BRI ISDN: BRI T-loops" on page 217
- — "Programming BRI S-loops, lines and ISDN devices" on page 223
- — "Configuring BRI lines" on page 219
- — "Configuring lines" on page 147
- — "Configuring lines: T1-Loop start" on page 187
- — "Configuring lines: T1-E&M" on page 181
- — "Configuring lines: T1-Digital Ground Start" on page 193
- — "Configuring lines: T1-DID" on page 199
- — "Configuring lines: PRI" on page 171
- — "Configuring VoIP lines" on page 414
- — "Configuring lines: DPNSS lines" on page 211
- — "Configuring lines: Target lines" on page 177
- — "Call Security: Configuring Direct Inward System Access (DISA)" on page 451

- Networking, private and public
  - — "Public networking: Setting up basic systems" on page 323
  - — "Public networking: Tandem calls from private node" on page 327
  - — "Private networking: Basic parameters" on page 349
  - — "Private networking: MCDN and ETSI network features" on page 353
  - — "Private networking: Using shared line pools" on page 347
  - — "Private networking: Using destination codes" on page 373
  - — "Private networking: PRI Call-by-Call services" on page 377
  - — "Private networking: PRI and VoIP tandem networks" on page 357
  - — "Private networking: MCDN over PRI and VoIP" on page 329
  - — "Private networking: DPNSS network services (UK only)" on page 365
  - — "Configuring centralized voice mail" on page 385
- Dialing plan configuration

Figure 13 Dialing plan configuration


Hardware installation and initial system properties config
System resources
Start DN/DN length
Network config
Keycodes
Public/Private dialing plans
Number plan configuration overview
Call control and telephony programming
MBMs/Lines and loops
CLID
Received # lengths (incoming)
Access codes (into and within system)
Destination codes (outgoing)
Remote access pkg. (incoming)
Restrictions (outgoing)
Remote system interoperability issues (outgoing)

— “Dialing plan: System settings” on page 303

— “Dialing plan: Public network” on page 311

— “Dialing plan: Line pools and line pool codes” on page 297

— “Dialing plan: Routing and destination codes” on page 289

— “Dialing plan: Routing configurations” on page 277

— “Configuring CLID on your system” on page 235

— “Call security: Restriction filters” on page 457 (outgoing calls)

— “Call security: Remote access packages” on page 463 (incoming calls)

Incoming calls

For incoming calls, you can have a central reception point, or you can specify target lines to one or more telephones to receive directed calling.

You can arrange your telephones in Hunt groups, ringing groups, or call groups that use system-wide call appearance (SWCA) assignments to share calls.

You can also configure lines for use by system users who call in from outside the system. You can give them direct access to the system with an Auto DN, or you can configure the line so they hear a stuttered dial tone, at which point they need to enter a password (CoS) to gain access (DISA DN).

Figure 14 Configuring incoming call traffic (network)

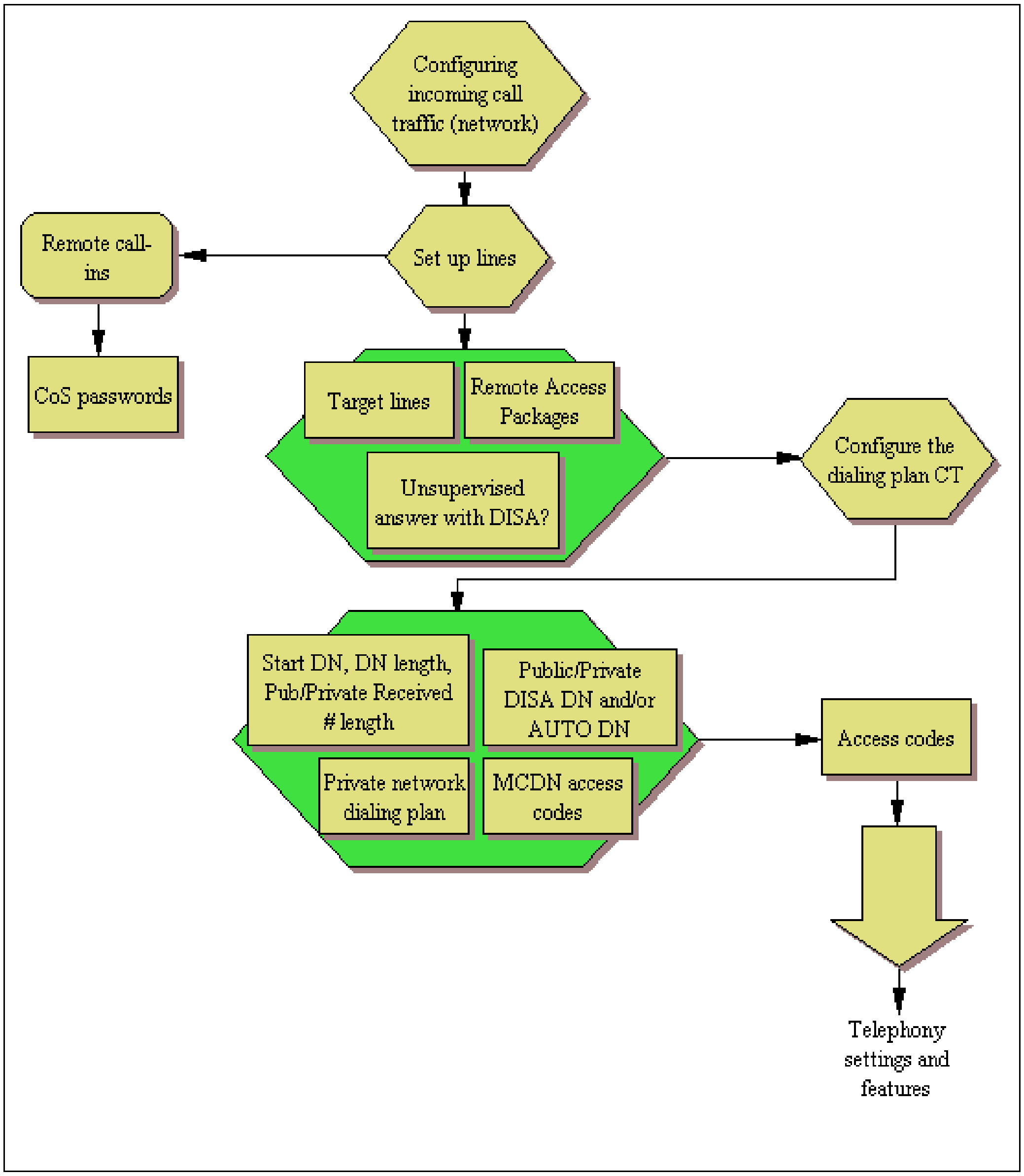

Configuring incoming call traffic (network)
Set up lines
Remote call-ins
CoS passwords
Target lines
Remote Access Packages
Unsupervised answer with DISA?
Configure the dialing plan CT
Start DN, DN length, Pub/Private Received # length
Public/Private DISA DN and/or AUTO DN
Private network dialing plan
MCDN access codes
Access codes
Telephony settings and features

Figure 15 Configuring incoming call traffic (telephones)

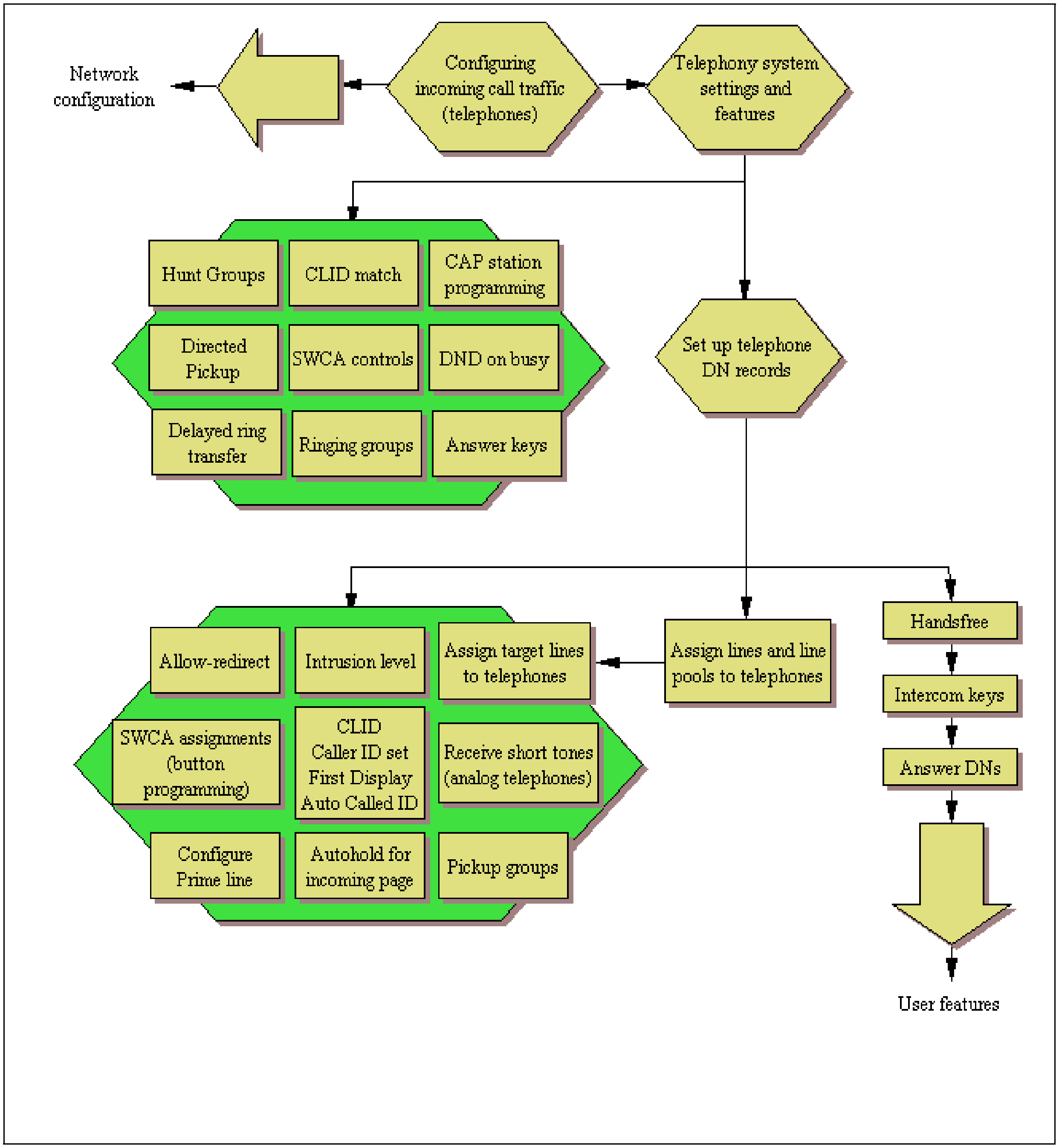
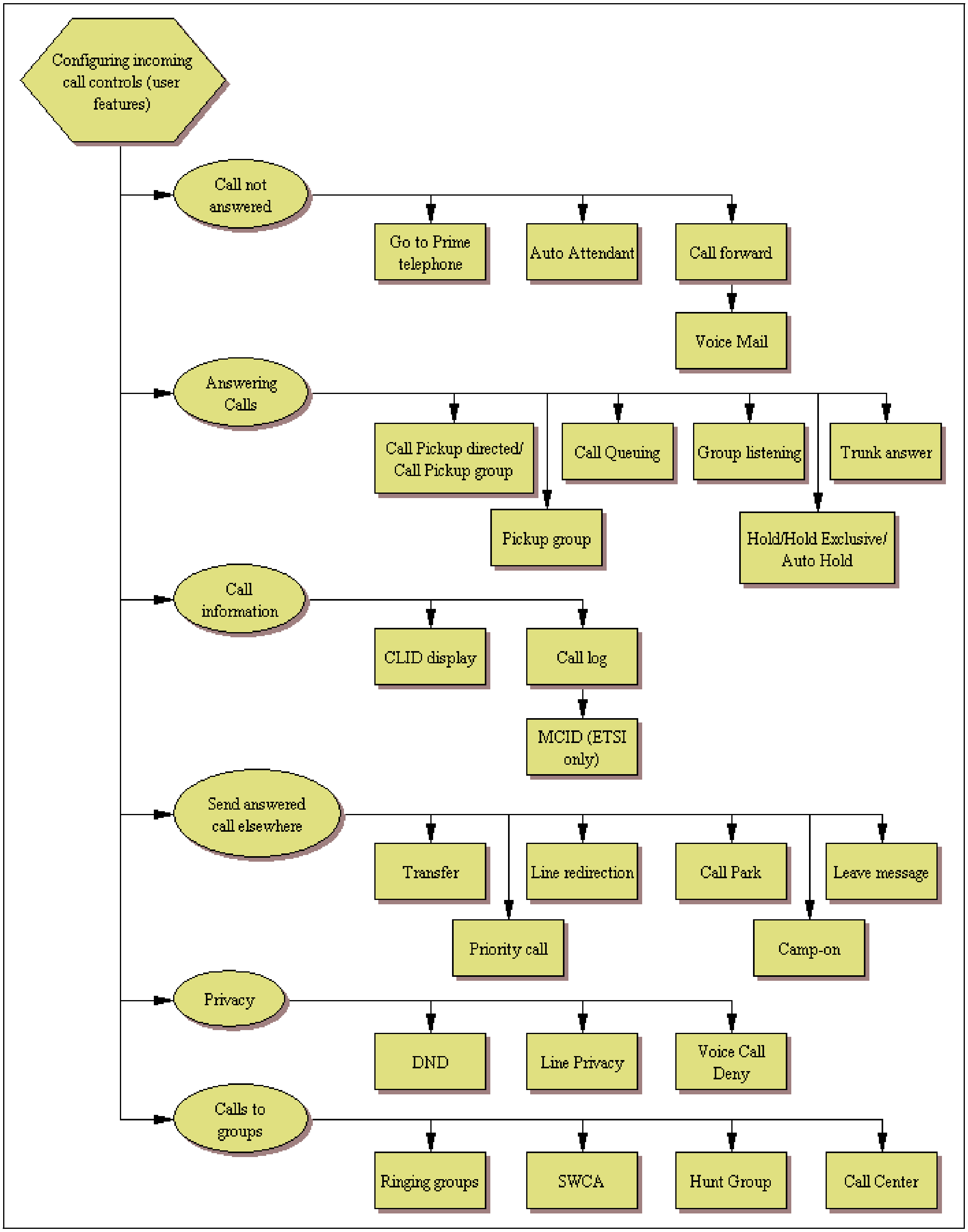

Network configuration
Configuring incoming call traffic (telephones)
Telephony system settings and features
Hunt Groups
CLID match
CAP station programming
Directed Pickup
SWCA controls
DND on busy
Delayed ring transfer
Ringing groups
Answer keys
Set up telephone DN records
Allow-redirect
Intrusion level
Assign target lines to telephones
Assign lines and line pools to telephones
Handsfree
Intercom keys
Answer DNs
SWCA assignments (button programming)
CLID Caller ID set First Display Auto Called ID
Receive short tones (analog telephones)
Configure Prime line
Autohold for incoming page
Pickup groups
User features

Figure 16 Configuring incoming call controls (user features)



Outgoing calls

For outgoing calls, you can assign one or more intercom keys to directly link to a line pool or prime line, or allow line pool access codes, destination codes, or internal system numbers to direct the call. Telephones without intercom keys on the telephone have intercom keys assigned, but the user must pick up the handset to access calls. In this case, the intercom key is an assigned DN.

For calls within the system, all telephones are virtually linked within the system. To call another telephone inside the system, lift the handset and dial the local DN. In this case, the prime line has to be set to intercom or none.

For calls going outside the system:

- If you assign the prime line to a line pool, all the lines in that line pool must be assigned to the telephone. When you pick up the handset, the telephone automatically grabs the first available line from the assigned line pool. In this configuration, you must ensure that the outgoing number is allowed by the line pool.
- If you assign the prime line to an intercom button, when you press the intercom button you get system dial tone. Then, you enter a line pool access code or a destination code to direct the outgoing call to the appropriate line pool, where it exits the system on any available line in that pool.

Figure 17 Configuring outgoing call traffic (Sheet 1 of 2)

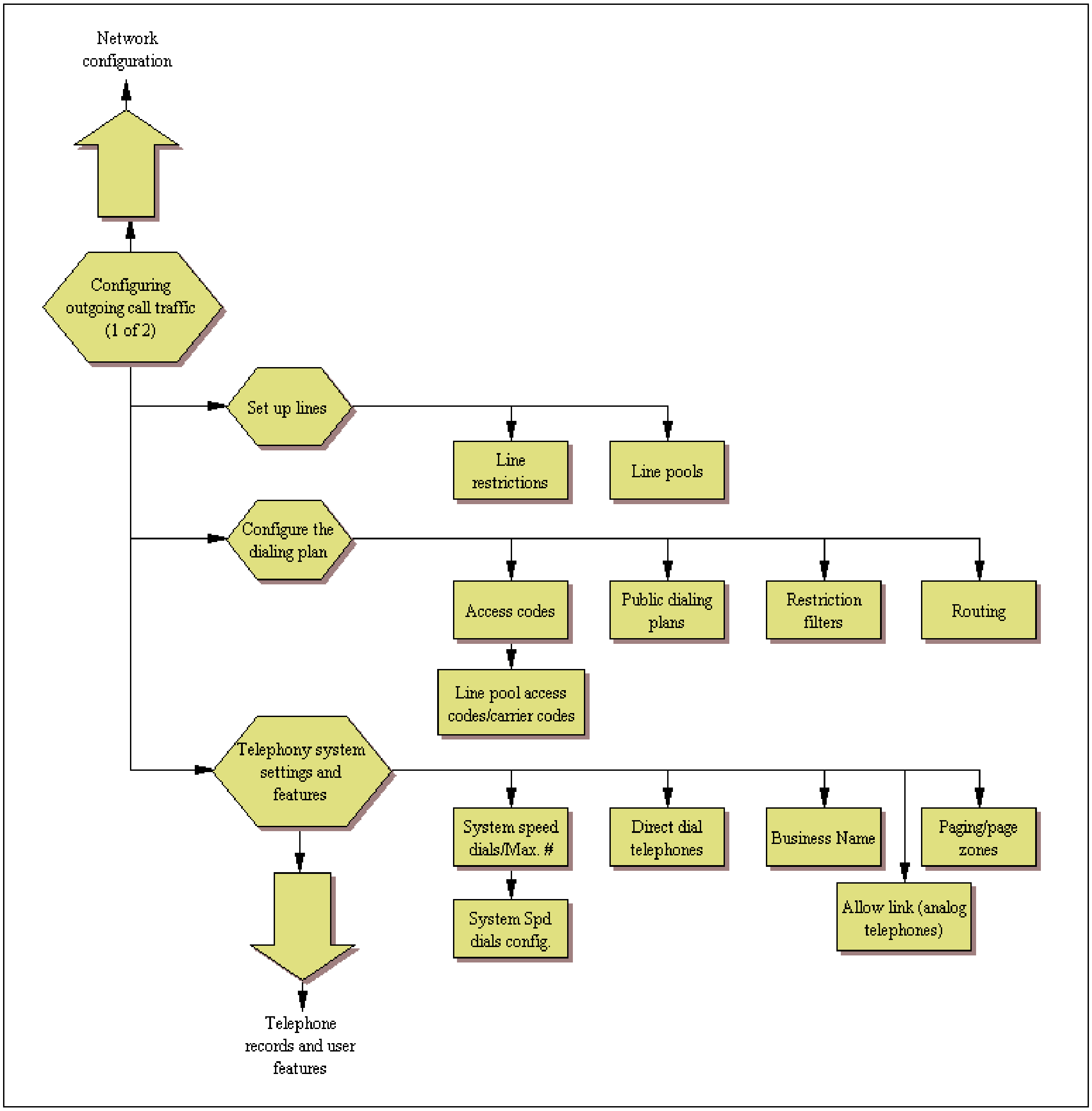
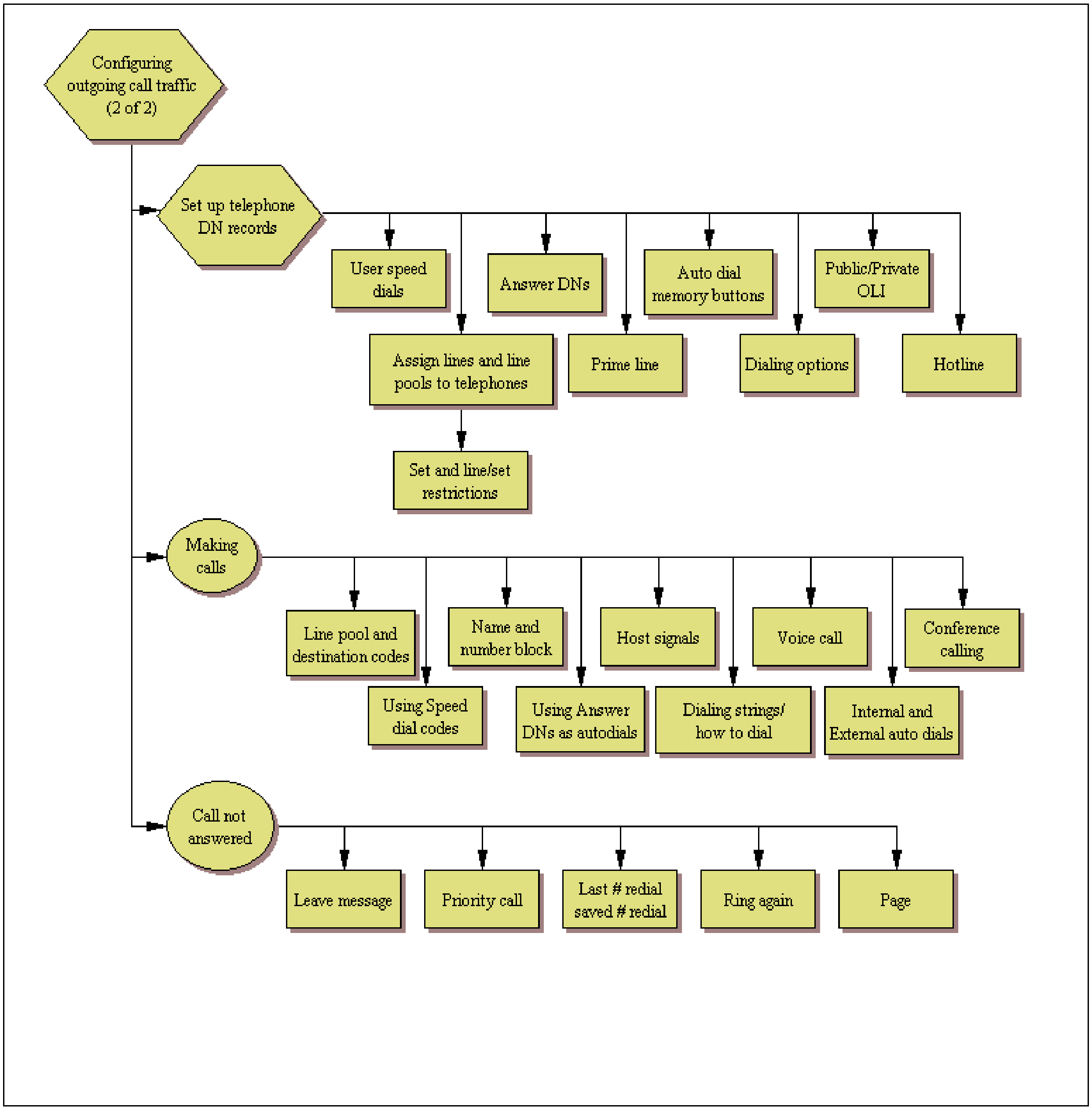

Network configuration
Configuring outgoing call traffic (1 of 2)
Set up lines
Line restrictions
Line pools
Configure the dialing plan
Access codes
Public dialing plans
Restriction filters
Routing
Line pool access codes/carrier codes
Telephony system settings and features
System speed dials/Max. #
Direct dial telephones
Business Name
Paging/page zones
System Spd dials config.
Allow link (analog telephones)
Telephone records and user features

Figure 18 Configuring outgoing call traffic (Sheet 2 of 2)


Configuring outgoing call traffic (2 of 2)
Set up telephone DN records
User speed dials
Answer DNs
Auto dial memory buttons
Public/Private OLI
Assign lines and line pools to telephones
Prime line
Dialing options
Hotline
Set and line/set restrictions
Making calls
Line pool and destination codes
Name and number block
Host signals
Voice call
Conference calling
Using Speed dial codes
Using Answer DNs as autodials
Dialing strings/ how to dial
Internal and External auto dials
Call not answered
Leave message
Priority call
Last # redial saved # redial
Ring again
Page

Chapter 4
Application resources panel

The application resources panel allows you to modify resources allocated to applications on the BCM. While the panel tracks four types of resources, DSP resources are generally the only type of resources that affect performance on the BCM. For more information on planning your application resources, see "Determining the resources you require" on page 86.

Note: Do not change these settings unless you want to restrict resources.

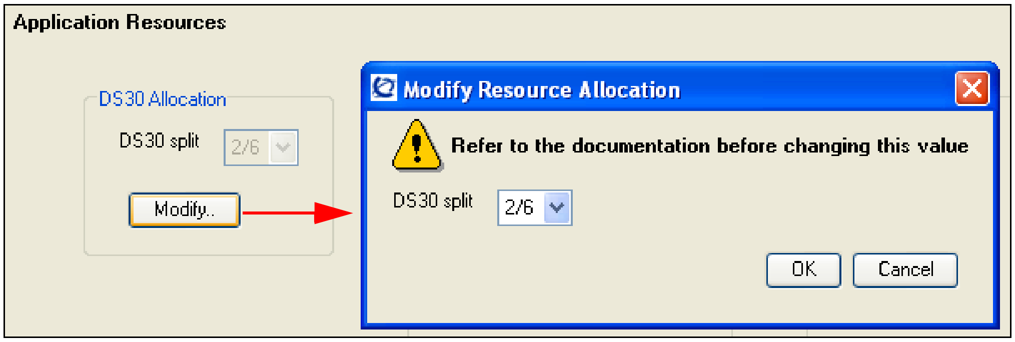
The application resources panel consists of three tables and a panel:

- DS30 Allocation
- "Total Resources" on page 76
- "Reserved Resources" on page 77
- "Application Resource Reservations" on page 95

DS30 Allocation

The DS30 allocation determines how many IP telephones you can connect to your system. If you have a system that does not use IP telephones, the number of signaling channels does not affect your configuration. For more information, refer to "Changing the DS30 split" on page 96.

Figure 19 DS30 Allocation



DS30 split 2/6

Refer to the documentation before changing this value

DS30 split 2/6

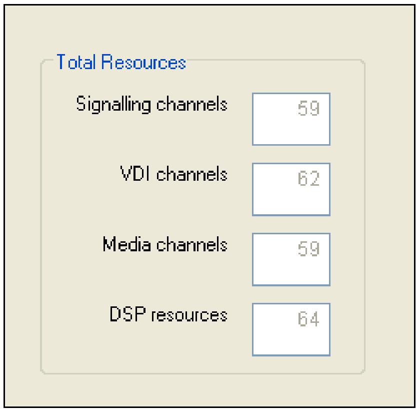
Table 2 DS30 allocation

Attribute	Value	Description
DS30 split	<read-only>	Number of signaling channels available.
Actions		
Modify	<button>	Modify the resource allocation.
DS30 split	2/6 3/5	Select the resource allocation.
OK	<button>	Confirm the selection. This may require a system reboot.
Cancel	<button>	Cancel change.

Total Resources

The total resources options show the maximum resources available for each type of resources.

Figure 20 Total Resources



Signalling channels 59

VDI channels 62

Media channels 59

DSP resources 64

Table 8 Total Resources (Sheet 1 of 2)

Attribute	Value	Description
Signalling channels	<read-only>	The total number of signalling channels on the system.

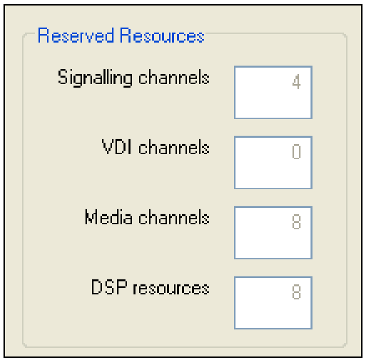
Table 8 Total Resources (Sheet 2 of 2)

Attribute	Value	Description
VDI channels	<read-only>	The total number of VDI channels on the system.
Media channels	<read-only>	The total number of media channels on the system.
DSP resources	<read-only>	The total number of DSP resources on the system.

Reserved Resources

The Reserved Resources options show the resources currently reserved or in use.

Figure 21 Reserved Resources



Reserved Resources

Signalling channels 4

VDI channels 0

Media channels 8

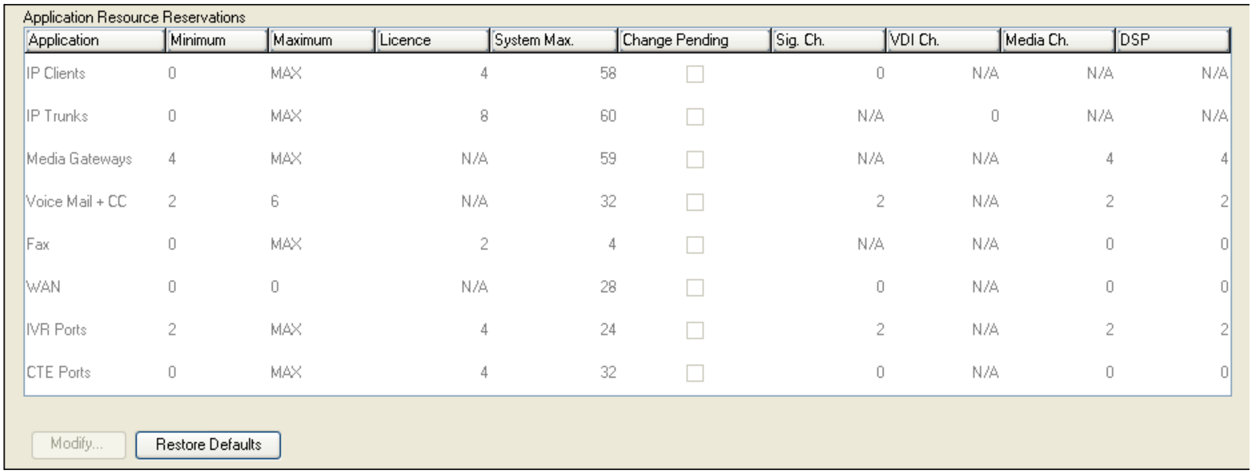
DSP resources 8

Table 9 Reserved Resources

Attribute	Value	Description
Signalling channels	<read-only>	The number of signalling channels in use on the system. This number can change based on the values entered for applications, and on the those applications currently in use.
VDI channels	<read-only>	The number of VDI channels in use on the system. This number can change based on the values entered for applications, and on the those applications currently in use.
Media channels	<read-only>	The number of media channels in use on the system. This number can change based on the values entered for applications, and on the those applications currently in use.
DSP resources	<read-only>	The number of DSP resources in use on the system. This number can change based on the values entered for applications, and on the those applications currently in use.

The Application Resource Reservations table allow you to set minimum and maximum values for each of six types of applications. The table contains 10 columns, eight of which are read-only. For information on determining the appropriate values for each type of application, see “Rules for managing the resources” on page 83.

Figure 22 Application Resource Reservations



Application Resource Reservations

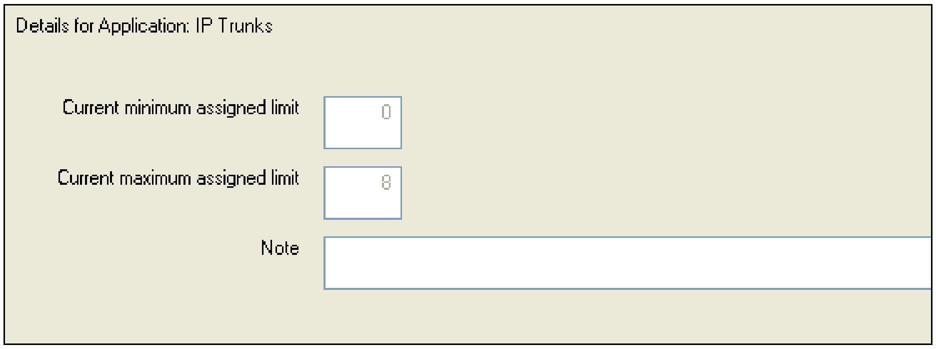
Application	Minimum	Maximum	Licence	System Max.	Change Pending	Sig. Ch.	VDI Ch.	Media Ch.	DSP
IP Clients	0	MAX	4	58	□	0	N/A	N/A	N/A
IP Trunks	0	MAX	8	60	□	N/A	0	N/A	N/A
Media Gateways	4	MAX	N/A	59	□	N/A	N/A	4	4
Voice Mail + CC	2	6	N/A	32	□	2	N/A	2	2
Fax	0	MAX	2	4	□	N/A	N/A	0	0
WAN	0	0	N/A	28	□	0	N/A	0	0
IVR Ports	2	MAX	4	24	□	2	N/A	2	2
CTE Ports	0	MAX	4	32	□	0	N/A	0	0

Table 10 Total Resources

Attribute	Value	Description
Application	<read-only>	The name of the application.
Minimum	Numeric value	The minimum number of resources reserved at all times for the application. If a value of 2 is entered, the system will always reserve enough resources for 2 instances of the application.
Maximum	Numeric value, or the string MAX	The maximum number of applications to allow. If the value is set to MAX, the system will allow up to the system maximum, as long as there are enough resources.
Licence	<read-only>	The number of licenses the system has activated for the application. If the value is N/A, the application does not require licenses.
System Max.		The maximum instances of an application the BCM can support.
Change Pending	<read-only>	If this box is checked, a change is pending to the system. Most changes take effect immediately, but in some instances, a change may wait until applications shut down. Details about changes pending can be seen in the details panel.
Sig. Ch.	<read-only>	The number of signalling channels reserved by the application. This can be changed by modifying the minimum and maximum values for the application. If the field has a value of N/A, the application does not require this type of resource.
VDI Ch.	<read-only>	The number of VDI channels reserved by the application. This can be changed by modifying the minimum and maximum values for the application. If the field has a value of N/A, the application does not require this type of resource.
Media Ch.	<read-only>	The number of media channels reserved by the application. This can be changed by modifying the minimum and maximum values for the application. If the field has a value of N/A, the application does not require this type of resource.
DSP	<read-only>	The number of DSP resources reserved by the application. This can be changed by modifying the minimum and maximum values for the application. If the field has a value of N/A, the application does not require this type of resource.

Details for application

The Details for Application panel changes whenever a different row is selected from the Application Resource Reservations table. It reflects the current minimum and maximum limits, in instances where changes do not happen immediately.



Details for Application: IP Trunks

Current minimum assigned limit 0

Current maximum assigned limit 8

Note

Table 11 Total Resources

Attribute	**Value**	**Description**
Current minimum assigned limit	<read-only>	The current minimum assigned for an application.
Current maximum assigned limit	<read-only>	The current maximum assigned for an application.
Note	text field	Enter any notes regarding these limits.

Chapter 5
Configuring application resources

The following describes how to set up the application resources controlled by the Media Services Card (MSC), which is the control center for voice and data traffic in the BCM.

The following path indicates where to configure the resources in Element Manager:

- Element Manager: **Configuration > Resources > Application Resources**

Warning: Only system administrators should have access to these Element Manager records. Changing settings can affect other parts of the system. You need to understand the consequences of any changes before you make them. Some changes are NOT reversible.

Refer to the following topics:

- "Types of resources"
- "Rules for managing the resources" on page 83
- "Determining the resources you require" on page 86
- "Understanding the minimum and maximum values" on page 94
- "Changing the DS30 split" on page 96

Types of resources

Application resources are required for the following features:

- system functions
- voice mail, contact center, and IVR (Interactive Voice Response)
- Fax mail
- IP telephony trunks
- IP clients
- Dial-on-Demand (DoD) WAN and Backup ISDN WAN connections

When you configure the resources, you are configuring how BCM shares the resources between these features.

There are several values that you must check when you are configuring resources:

- "Signaling channels" on page 82
- "Media channels" on page 82
- "DSP resources" on page 82
- "Voice bus paths" on page 82
- "Media gateways" on page 82

Signaling channels

Signaling channels are the communication channels used to send control signals to and from the MSC. You must have one signaling channel for each connected device and enabled feature port.

The number of signaling channels on your system determines the number of devices that you can connect and feature ports that you can enable on your system. Signaling channels are also known as D-channels.

Media channels

Media channels are the communication channels used to send voice and data information between the devices and feature ports. Media channels are required only when a device or feature is sending or receiving voice or data information. For this reason, the devices and feature ports can share media channels.

The number of media channels you have determines how many devices and feature ports can exchange voice and data information at the same time. Media channels are also known as B-channels.

DSP resources

Digital Signal Processors (DSP) provide the voice processing functions on BCM. Voice processing is required to convert voice information to and from digital format for voice mail, Contact Center and IVR. Voice processing is also required to handle encoding and decoding of IP telephony calls. The DSPs are located on the MS-PEC cards installed in your MSC.

The number of DSP resources you have determines the number of voice mail ports, contact center ports, Fax mail ports, IVR ports, IVR Fax ports, WAN connections, and IP telephony calls that can be active at the same time.

Voice bus paths

The voice bus paths are the communication channels between the DSPs on the MS-PECs and the master DSP on the MSC. One voice bus path is required for each voice processing task that is operating on the DSPs.

There are 62 voice bus paths available on BCM.

Media gateways

Media gateways are logical connections that are a combination of DSP resources, media channels, and voice bus paths that provide protocol translation between IP telephones and trunks and analog and digital telephony devices.

Rules for managing the resources

The following rules are provided to assist you in configuring your resources.

- "Signaling channel rules"
- "Media channel rules" on page 83
- "DSP resources rules" on page 84
- "Voice bus path" on page 85
- "Media gateways" on page 85

Signaling channel rules

Signaling channels determine how many IP telephones you can connect to your system. If you have a system that does not use IP telephones, the number of signaling channels does not affect your configuration.

- The total number signaling channels available depends on the DS30 split you have configured. For information about how to view and change the DS30 split, refer to "Changing the DS30 split" on page 96.
  If you have a 2/6 DS30 split, the total number of signaling channels is 64.
  If you have a 3/5 DS30 split, the total number of signaling channels is 96.
- Management functions use six signaling channels.
- Dial-on-Demand ISDN WAN uses 27 signaling channels.
  All 27 signaling channels are used, regardless of the number of WAN channels configured.
- Voice Mail requires one signaling channel for each voice mail port enabled. You can enable up to 32 voice mail ports.
  Both voice mail and contact center use Voice Mail ports.
- IP Telephony clients require one signaling channel for each IP telephone connected to the system.
- IP Telephony trunks require one signaling channel.
  Only one signaling channel is required regardless of the number of IP Telephony trunks enabled.
- IVR requires 1 signaling channel for each enabled IVR port.
- Up to 24 ports enabled. Maximum of 32 ports between IVR and voice mail.

Media channel rules

The media channels are used to transport voice and data signals between devices.

- Management functions use five media channels. These five channels are reserved for management functions and are always in use.
- Dial-on-Demand ISDN WAN uses 27 media channels.
  All 27 media channels are used, regardless of the number of WAN channels configured. The maximum number of WAN channels is 16.
- Voice Mail and contact center use one media channel for each active session.

- A call between an IP telephone and a digital or analog telephone or a PSTN line uses a media channel for the duration of the call.
- A call from a digital or analog telephone that uses an IP trunk uses a media channel for the duration of the call.
- A call between two IP telephones on the same BCM uses a media channel during call setup. After the call is established, the media channel is released.
- A call on an IP telephone using an IP trunk uses a media channel during call setup. After the call is established, the media channel is released.
- IVR needs 1 media channel for each active session.

Since most of the devices do not use media channels all of the time, your system can have more devices than there are media channels. However, to ensure you have sufficient system resources, make sure the number of media channels you have exceeds your estimate of peak media channel usage. The section below provides an example of how to estimate your peak media channel usage.

Example of how to estimate peak media channel usage

The example below is for a fictional company named CompanyABC. The numbers used are strictly for this example. Actual numbers will vary, depending on the company. When you are estimating your peak media channel usage, make sure you use numbers that reflect your business.

- CompanyABC has a BCM system with 96 telephones. Of these telephones, 48 are digital telephones and 48 are IP telephones.
  The percentage of IP telephones is 50% (48/96). This percentage is used to estimate how many calls will be made between IP telephones and digital telephones.
- In CompanyABC, the users are typically on the telephone 15 minutes out of each hour, or 25% of the time. During peak hours, the users are on the telephone 30 minutes, or 50% of the time. Therefore, the peak usage of IP telephones is 24 (50% X 48 IP telephones).
- In CompanyABC, half of the calls are made to external destinations and half of the calls are made within the BCM system. CompanyABC does not have IP trunks, so the calls from the IP telephones to external destinations must use PSTN lines.
  The peak number of IP telephone calls that use PSTN lines is 12.
  (50% of calls external X 24 IP telephones during peak usage.)
- For internal calls, there is a 50% chance the call is made to a digital telephone.
  The peak number of IP telephone calls to digital telephones is 6.
  (50% of calls internal X 24 IP telephones peak usage X 50% number of digital telephones.)
- The peak media channel usage for IP telephony is 18.
  (12 media channels for external calls and 6 for calls made to digital telephones.)

DSP resources rules

The number of DSP resources you have depends of the number and type of MS-PECs that are installed.

For the purposes of calculating DSP resources, we can estimate the relative power of each configuration as follows:

• 4 MS-PEC I	24 units
• 2 MS-PEC III	64 units
• 4 MS-PEC III	128 units

The number of DSP resources you need depends on the features and type of codec you are using. Refer to Appendix B, "Codec rates," on page 713.

- Dial-on-Demand WAN uses 1 unit for each 64Kbit/s channel.
- Voice Mail, IVR, and contact center use 1 unit for each active session.
- Fax uses 6 units for each active fax channel.
- IP telephone or IP trunk using G.711 codec uses 1 unit.
- IP telephone or IP trunk using G.729 codec uses 3 units.
- IP telephone or IP trunk using G.723 codec uses 4 units.

Note: Some of the DSP resource units in the preceding list are rounded to the nearest whole number. This is done to ease the calculation of the DSP resources you require. To calculate more accurate DSP requirements, use the DSP resource units in shown in the following table.

Table 12 DSP resource requirements

Feature or codec	Resource units on an MS-PEC I	Resource units on an MS-PEC III
G.729	3	2.75
G.723	4	4.2
Fax	5	6
T.38 IP Fax	5	6
IVR Fax	6	6

Voice bus path

There are 62 voice bus paths available on BCM.

- Voice mail and IVR use one voice bus path for each active session.
- Dial-on-Demand WAN uses one voice bus path for each 64Kbit/s channel that is active.
- IP telephones and IP trunks require one voice bus path when ever a media channel is required.

Media gateways

One media gateway is required for each call:

- from an IP telephone to an analog or digital telephone
- from an IP telephone using a PSTN line
- from an analog or digital telephone using an IP trunk

Determining the resources you require

The following questions are designed to help determine how many resources you require. Based on the answers to these questions, you can calculate the number of signaling channel*s*, media channels, voice bus paths, and DSP resource units you need. Use the table in "Evaluation" on page 92 to determine the configurations.

Note: In the following questions, "peak periods" refers to the periods of time when there is the highest overall activity. It is necessary to consider the resource requirements for "peak periods" to determine if available voice bus paths and DSP resources meet your resource requirements at all times.

ISDN WAN (Dial-up/Nailed-up)

As you answer the following questions, record your answers in the table in "Record of required resources" on page 91.

1 What is the maximum required WAN bandwidth?
The range is 0 to 1 Mbit/s (16 x 64 kbit/s) in 64 kbit/s increments.
If the answer is more than zero:

- Add 27 to the signaling channel count.
- Add 27 to the media channel count.

2 What is the required WAN bandwidth during peak periods?
The range is 0 to the maximum bandwidth you entered in question 1.
For each 64 kbit/s of bandwidth:

- add 1 to the voice bus time slot count
- add 1 to the DSP resource unit count

Voice mail and CC

3 What is the maximum number of voice mail ports required? Voice mail ports are used for voice mail and Contact Center.
The range is 0 to 32 ports.

For each voice mail port:

- add 1 to the signaling channel count
- add 1 to the media channel count

4 What is the number of Voice mail ports required during peak periods?
The range is 0 to the maximum number of ports selected in question 3.

For each voice mail port

- add 1 to the voice bus path count
- add 1 to the DSP resource unit count

5 How many fax tasks will be used during peak periods?
The range is 0 to 2.

For each fax task:

- add 6 to the DSP resource unit count

Note: The maximum number of voice ports shared between voice mail and IVR is 32. The maximum number of ports shared between voice mail, IVR, and T.38 IP fax is eight. There are only 2 fax ports.

Note: The fax DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

IVR and IVR Fax

6 What is the maximum number of IVR ports required? IVR ports are used for interactive voice response applications.
The range is 0 to 24 ports.

For each voice mail port:

- add 1 to the signaling channel count
- add 1 to the media channel count

7 What is the number of IVR ports required during peak periods?
The range is 0 to the maximum number of ports selected in question 6.
For each voice mail port:

- add 1 to the voice bus path count
- add 1 to the DSP resource unit count

8 How many fax tasks will be used during peak periods?
The range is 0 to max.

For each fax task:

- add 6 to the DSP resource unit count

Note: The maximum number of voice ports shared between voice mail and IVR is 32. The maximum number of ports shared between voice mail, IVR, and T.38 IP Fax is eight. There are only 2 fax ports.

Note: The fax DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

IP telephones

9 What is the maximum number of IP telephones required?
The range is 0 to 90 IP telephones.

For each IP telephone:

- add 1 to the signaling channel count

10 How many IP telephones will be calling an analog or digital telephone or using a PSTN trunk during peak periods?
The range is 0 to the maximum number of IP telephones selected in question 9.

For each IP telephone:

- add 1 to the media channel count
- add 1 to the voice bus path count

11 How many IP telephones specified in question 10 will be using the G.711 codec?
The range is 0 to the maximum number of IP telephones selected in question 10.

For each IP telephone:

- add 1 to the DSP resource unit count

12 How many IP telephones specified in question 10 will be using the G.729 codec?
The range is 0 to the maximum number of IP telephones selected in question 11.

For each IP telephone:

- add 3 to the DSP resource unit count

Note: The G.729 DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

13 How many IP telephones specified in question 10 will be using the G.723 codec?
The range is 0 to the maximum number of IP telephones selected in question 11.

For each IP telephone:

- add 4 to the DSP resource unit count

Note: The G.723 DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

IP Trunks

14 What is the maximum number of IP trunks required?
The range is 0 to 60 IP trunks.

If the number is greater than zero IP trunks:

- add 1 to the signaling channel count

15 How many analog or digital telephones (not IP telephones) will use IP trunks during peak periods?
The range is 0 to the maximum number of IP trunks selected in question 14.

For each IP trunk:

- add 1 to the voice bus path count
- add 1 to the media channel count

16 How many IP trunks specified in question 15 will be using the G.711 codec?
The range is 0 to the maximum number of IP trunks selected in question 15.

For each IP trunk:

- add 1 to the DSP resource unit count

17 How many IP trunks specified in question 16 will be using the G.729 codec?
The range is 0 to the maximum number of IP trunks selected in question 16.

For each IP trunk:

- add 3 to the DSP resource unit count

Note: The G.729 DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

18 How many IP trunks specified in question 16 will be using the G.723 codec?
The range is 0 to the maximum number of IP trunks selected in question 16.
For each IP trunk:

- add 4 to the DSP resource unit count

Note: The G.723 DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

19 How many T.38 fax tasks will be used during peak periods?
The range is 0 to 8.

For each fax task:

- add 6 to the DSP resource unit count

Note: The maximum number of ports shared between voice mail, IVR, and T.38 IP Fax is eight. There are only 2 fax ports.

Note: The fax DSP resource unit count is rounded to ease calculations. For a more accurate DSP resource unit count, refer to the table in "DSP resources rules" on page 84.

Note: If the source or destination of the T.38 IP Fax can be fax mail or IVR, the fax message requires two fax tasks (12 units). One fax task handles the IP Fax portion of the transmission, and the other task handles the IVR or fax mail portion of the transmission.

Note: To use T.38 Fax, you must have 2 or 4 MS-PEC III installed in your MSC card.

Record of required resources

Use Table 13 to record the resources you require for your BCM system. To determine the resources that you require, answer the questions in “Determining the resources you require” on page 86.

Table 13 Required resources

Question	**Answer**	**Signaling channels**	**Media channels**	**Voice bus paths**	**DSP resource units**
1. WAN				---	---
2. Peak WAN		---	---		
3. VM/ACD				---	---
4. IVR					
5. Peak VM/ACD		---	---		
6. Peak FAX		---	---	---	
7. Peak IVR					
8. IVR FAX					
9. IP Sets			---	---	---
10. Peak IP Sets		---			---
11. IP Sets G711		---	---	---	
12. IP Sets G729		---	---	---	
13. IP Sets G723		---	---	---	
14. IP Trunks			---	---	---
15. Peak IP Trunks		---			---
16. IP Trunks G.711		---	---	---	
17. IP Trunks G.729		---	---	---	
18. IP Trunks G.723		---	---	---	
19. IP Trunks T.38 Fax		---	---	---	
Totals					

Evaluation

After you answer the questions and calculate the four totals, use the following rules to determine the required BCM configuration.

Table 14 Evaluation of required BCM configuration

Resource	Number required	Required configuration
Signaling channel count	58 or less	2/6 DS30 split
	59 to 90	3/5 DS30 split
	91 or more	exceeds BCM capacity
Media channel count	58 or less	2/6 DS30 split
	59 to 90	3/5 DS30 split
	91 or more	exceeds BCM capacity
Voice bus path count	62 or less	within BCM capacity
	63 or more	exceeds BCM capacity
DSP resource units	1 to 24	4 MS PEC I
	1 to 64	2 MS PEC III
	65 to 128	4 MS PEC III
	129 or more	exceeds BCM capacity

➡ **Note:** If your system requires more resources than are available on your MS-PEC configuration, you can upgrade your MS-PECs. For information about how to upgrade your MS-PECs, refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

Example of a BCM configuration

The following two tables provide examples of required configurations.

Table 15 Example of required configuration

Question	Answer	Signaling channels	Media channels	Voice bus paths	DSP resource units
1. WAN	512 kbit/s (8)	27	27	---	---
2. Peak WAN	512 kbit/s (8)	---	---	8	8
3. VM/ACD	8	8	8	---	---
4. IVR					
5. Peak VM/ACD	6	---	---	6	6
6. Peak IVR					
7. Peak FAX	1	---	---	---	6
8. IVR FAX					
9. IP Sets	24	24	---	---	---
10. Peak IP Sets	12	---	12	12	---
11. IP Sets G711	6	---	---	---	6
12. IP Sets G729	4	---	---	---	12
13. IP Sets G723	2	---	---	---	8
14. IP Trunks	32	1	---	---	---
15. Peak IP Trunks	20	---	20	20	---
16. IP Trunks G.711	12	---	---	---	12
17. IP Trunks G.729	6	---	---	---	18
18. IP Trunks G.723	2	---	---	---	8
19. IP Trunks T.38 Fax		---	---	---	
Totals	---	60	67	46	84

Table 16 Evaluation for the example of required configuration

Resource	Number required	Recommended configuration
Signaling channel count	60	3/5 DS30 split
Media channel count	67	3/5 DS30 split
Voice bus path count	46	within BCM capacity
DSP resource units	84	4 MS-PEC III

Understanding the minimum and maximum values

The MSC Configuration allows you to determine how the resources are assigned on your BCM.

In some BCM systems, the total number of features and devices that require resources exceeds the number of resources that are available. To address this issue, BCM allows you to share the resources. By changing minimum and maximum values for each component you can fine tune this sharing.

Minimum

The minimum value is the number of resources that are always assigned to a component. You use this number to ensure a base level of service for a specific component. For example, to ensure that at least four people can be using voice mail at all times, you would enter four as a minimum value for the Voice Port component.

The resources that are not assigned using the minimum values are shared by the components. If a component needs additional resources, it can use some of the shared resources to provide service during the busy period. This method of sharing resources allows your BCM system to adapt to the changing demands for services.

Maximum

The maximum value is the maximum number of resources that a component can use. You use this number to ensure a single component does not consume all of the shared resources.

The MSC configuration you choose greatly affects the performance of your BCM system. Make sure you consider the needs of your users, including peak usage times, when selecting the minimum and maximum values. Table 17 describes the advantages and disadvantages of changing these values.

Table 17 Advantages and Disadvantages of Minimum and Maximum values (Sheet 1 of 2)

Value	Advantage	Disadvantage
Increasing Minimum Value	Increases the guaranteed level of service for a component. The DSP resources you assign as a Minimum are always available to the users of this component.	Decreases the flexibility of DSP resource sharing. DSP resources that are assigned to the Minimum value are not shared with other components. If you set the Minimum level too high, other components may not be available due to a lack of available DSP resources.
Decreasing Minimum Value	More DSP resources are available to share with other components. When there is a large pool of shared DSP resources, BCM more readily adapts to changing component use.	Lower guaranteed level of service for this component. If the Minimum value is too low, it is possible that some users will not be able to access this component when other components are in heavy use.
Increasing Maximum Value	Allows this component to use more of the shared DSP resources during times of peak use. This allows more people to use this component at the same time.	During times of peak use, this component may consume all of the shared resources. This may cause other components to be unavailable to users.

Table 17 Advantages and Disadvantages of Minimum and Maximum values (Sheet 2 of 2)

Value	Advantage	Disadvantage
Decreasing Maximum Value	Prevents this component from using so many of the shared DSP resources, that other components are unavailable.	Limits the number of people that can use this component even if sufficient DSP resources are available.

Application Resource Reservations

Table 18 describes each component on the MSC card.

Table 18 MSC custom configuration parameters (Sheet 1 of 2)

Component	Description
IP Clients	IP Clients are Nortel IP telephones. DSP resources are required only when the IP telephone is in use (for example, to make a call, receive a call, listen to voice mail). For information about how to configure IP clients, refer to the *BCM 4.0 Telephony Device Installation Guide* (N0060609) **Note**: The codec (G.711, G.723 or G.729) you are using for the IP Client affects how many IP clients you can use on your system.
IP Trunks	IP Trunks are communication channels that BCM uses to send and receive IP telephony calls using the Public Data Network. You can use IP trunks to connect your BCM system to: • another BCM system • a Meridian 1 IPT system For information about how to configure IP trunks, refer to the *BCM 4.0 Telephony Device Installation Guide* (N0060609). **Note**: The codec (G.711, G.723 or G.729) you are using for the IP Trunk affects how many IP Trunks you can use on your system.
Media Gateways	Media Gateways provide the connection between IP telephony devices (IP trunks, and Nortel IP telephones) and normal telephony devices (PSTN lines; 74XX, 7316E, 7316, 7208s, 7100, 7000 digital phones; analog telephones etc.).
Voice Mail + contact center	Voice Mail and contact center ports are communication channels that connect users to the CallPilot Voice Mail and Contact Center Software. DSP resources are required only when a user connects to voice mail or Contact Center. This includes callers hearing greetings, callers leaving messages, and users accessing their mailboxes. The minimum value for Voice Mail and Contact Center Ports must be 2 or higher, unless you want to disable CallPilot Voice Mail and Contact Center Software. The maximum value for Voice Mail and Contact Center Ports must be 2 or higher, unless you want to disable CallPilot voice mail and Contact Center Software. To disable CallPilot voice mail and Contact Center Software, change the minimum and maximum values for Voice Mail and Contact Center Ports to zero.
Fax	Fax ports are communication channels that connect a fax machine to the BCM. Fax mail ports are communication channels that connect a fax machine to a fax mailbox or a user to a Fax-on-Demand mailbox. IVR fax ports are communication channels that connect a fax machine to IVR functions. T.38 IP Fax ports are communication channels that connect to a fax machine that is using an IP trunk.
WAN	WAN channels are dialup ISDN WAN connections.

Table 18 MSC custom configuration parameters (Sheet 2 of 2)

Component	Description
IVR Ports	IVR ports are communication channels that connect users to the IVR Software. DSP resources are required only when a user connects to IVR. This includes callers hearing greetings, callers leaving messages, and users accessing their mailboxes. The minimum value for IVR Ports must be 2 or higher, unless you want to disable IVR. The maximum value for IVR Ports must be 2 or higher, unless you want to disable IVR. To disable IVR, change the minimum and maximum values for IVR Ports to zero.
CTE Ports	CTE ports are communication channels that connect CTE applications to the BCM. An example of a CTE application is BCM Personal Call Manager.

Changing the DS30 split

A DS30 bus is a group of 32 signaling channel and 32 media channels. The DS30 split determines how these channels are assigned on BCM.

You have a choice of a 2/6 or a 3/5 split. If you choose a 2/6 split, two DS30 buses are assigned to the MSC and six are assigned to the Media Bay Modules. If you choose a 3/5 split, three DS30 buses are assigned to the MSC and five are assigned to the Media Bay Modules.

The split you choose is determined by the number of signaling channels you require for applications such as voice mail, IVR, IP trunks, IP telephones, and dialup ISDN WAN connections. If you need 58 signaling channels or less for these applications, use a 2/6 DS30 split. If you need 59 signaling channels or more, use a 3/5 DS30 split.

If your signaling channel requirements change, for example you want to increase the number of IP telephones, you can change from a 2/6 setting to a 3/5 setting without losing data. All new records added after the update will reflect the new default settings. To determine what the channel requirements are, refer to “Determining the resources you require” on page 86.

⚠ **Warning:** Ensure that the system is idle before you perform this procedure. You must restart the system after you have changed the setting.

➡ **Note:** Ensure you have a current backup before you perform this procedure.

➡ **Note:** You must ensure that your system has adequate DSP resources to support an increase in voice processing traffic. To determine if you have enough DSP resources, refer to “Determining the resources you require” on page 86. If you need to add MS-PEC IIIs, refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) for installation instructions. Refer to the BCM sales catalogue for part numbers and ordering instructions.

Warning: If you choose to change the DS30 split of your system after you configure your system, you could risk losing data for both the core system and optional applications.

Make sure you understand the implications of the changes before you go forward with this procedure.

To change the DS30 split setting

1. Click **Configuration > Resources > Application Resources**.
2. Click **Modify** in the DS30 Allocation panel.
   A dialog box appears. The DS30 Split field displays the current setting for your system.
3. If you want to change the setting, choose the other option from the drop-down list.
4. Click **OK** to accept a change or click **Cancel** to leave the setting in the original state.
   If you click **OK**, you are prompted to restart the BCM server.

Note: Changing the DS30 split from 2/6 to 3/5 preserves the existing telephony data. Any new device records will have default data.

Any change in DS30 split requires a restart of the BCM for the change to be applied.

Chapter 6
Configuring resources — media bay modules

The following describes the Element Manager headings that define and control the settings for the media bay modules installed on your system.

The following paths indicate where to access the media bay modules in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Resources > Telephony Resources**
- Telset interface: ****CONFIG > Hardware**

Task:

Check settings for the media bay modules installed in the system.

Trunk modules:

- Confirm that the DIP switch setting matches the intended DS30 bus placement.
- Verify that the module type and programmed bus type settings under the intended DS30 bus are correct for the type of module installed.
- Configure the module parameters of the individual modules installed on each DS30 bus.

Station modules:

- Confirm that the DIP switch setting matches the intended DS30 bus placement.
- Verify that the module type and programmed bus type settings under the intended Bus # are correct for the type of module installed.

Note: Data and split-telephony/data module configuration are described in "Data modules" on page 519.

Refer to the following topics:

- "Explaining the Media Bay Modules headings" on page 100
- "Defining trunk module types and settings" on page 106
- "Internally-driven channels" on page 115
- "Working with the modules" on page 116

Media bay modules provide the BCM with physical interfaces to trunk (CO) lines and your system telephones, which are defined by directory number (DN) records. When media bay modules are first installed in your system, you need to configure them using the procedures described in this section. Media bay module DIP switch settings and installation procedures are described in the *BCM50 Installation and Maintenance Guide* (N027152) and the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

Also refer to "Changing the DS30 split" on page 96 for a description of the 2/6 and 3/5 split, which determines how many channels are available for media bay modules.

Explaining the Media Bay Modules headings

The **Telephony Resources** panel allows you to view and change settings for each media bay module installed in BCM.

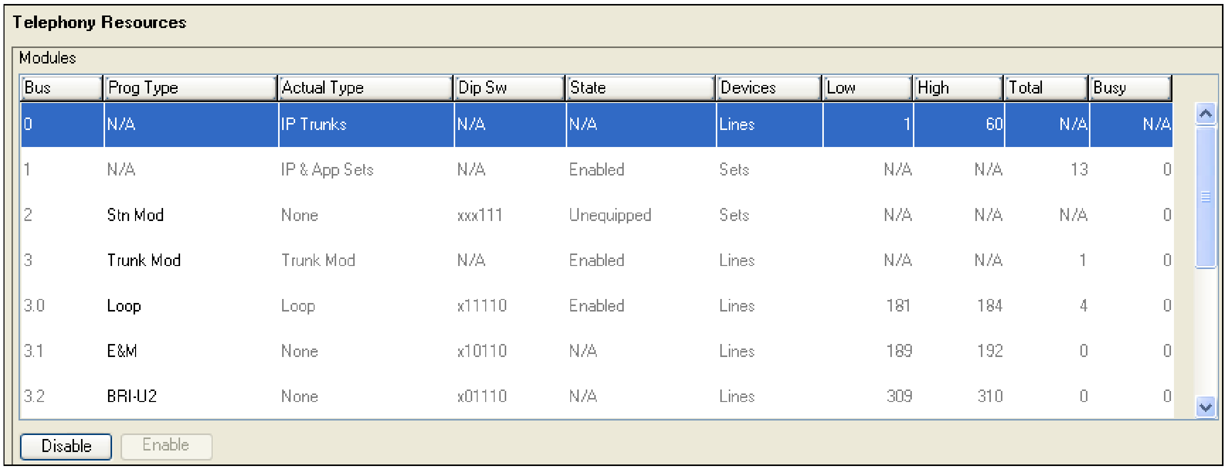
When you choose a region during your system startup, the Business Communications Manager installs a default set of media bay module settings. However, these may not be the settings that you want for the modules you install. Therefore, when you install a module, you must access the appropriate record and verify the settings for the module you installed.

Refer to the following:

- "Media bay module Bus numbers" on page 101
- "Identifying the module" on page 101
- "Module types and capacities" on page 104
- "Ports on Bus" on page 106

The following figure illustrates the column headings on the Telephony Resources panel. The exact items displayed in the Bus column depend on the type of module configured for that DS30 number.

Figure 23 Telephony Resources Modules panel



Bus	Prog Type	Actual Type	Dip Sw	State	Devices	Low	High	Total	Busy
0	N/A	IP Trunks	N/A	N/A	Lines	1	60	N/A	N/A
1	N/A	IP & App Sets	N/A	Enabled	Sets	N/A	N/A	13	0
2	Stn Mod	None	xxx111	Unequipped	Sets	N/A	N/A	N/A	0
3	Trunk Mod	Trunk Mod	N/A	Enabled	Lines	N/A	N/A	1	0
3.0	Loop	Loop	x11110	Enabled	Lines	181	184	4	0
3.1	E&M	None	x10110	N/A	Lines	189	192	0	0
3.2	BRI-U2	None	x01110	N/A	Lines	309	310	0	0

Tips: Some modules are region-based. If your system does not have the correct region installed during setup, the modules will not work. Refer to "Media bay module availability" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Note: Dimmed fields are read-only and cannot be changed.

Note: If you receive the error message `Telephony programming is currently not available. Please try again later.` when you click one of the column headings, this means that the part of the system that handles MSC is performing a reset. Wait 1and1/2 minutes and try again.

Media bay module Bus numbers

Under the headings for DS30 2 to 7 (or 2 to 6 if your system has a 3/5 DS30 split):

- Station or analog station modules display the **Ports on Module** heading.
- Trunk modules display from one to four **Module <#>** headings. These modules correspond to the offset configured on the module. A **Ports on Module** heading also appears for some types of modules (DTM set to PRI, and the BRI modules).
- If a WAN board is installed in the base unit, DS30 08 does not appear. Bus 08 and Bus 01 are used for internal media channels. If your system is set to a 3/5 split, DS30 07 is also used for media channels.

Identifying the module

Use these steps to define a Programmed Type. This setting notifies the Element Manager about what type of module is installed on the DS30 bus.

1 Click **Configuration > Resources > Telephony Resources**.

2 Click a **Bus** (2 to 7).
The details panel appears.

Note: Bus number is determined by the DS30 number set on the DIP switches of the module before it was installed.

3 Ensure the entry in the **ProgType** field agrees with the **Actual Type** of module that is installed for the DS30 bus, as shown Figure 24. Refer to "Module types and capacities" on page 101.

Programming tips: If the **Actual Type** reads **None**, choose the correct setting in the **Prog Type** field. After the system initializes to the module, the **Actual Type** should change to the correct module type. You may also have to disable, then re-enable the module to force the system to re-initialize (click **Disable** or **Enable** under the **Modules** table).

Some modules take a few minutes to reinitialize.

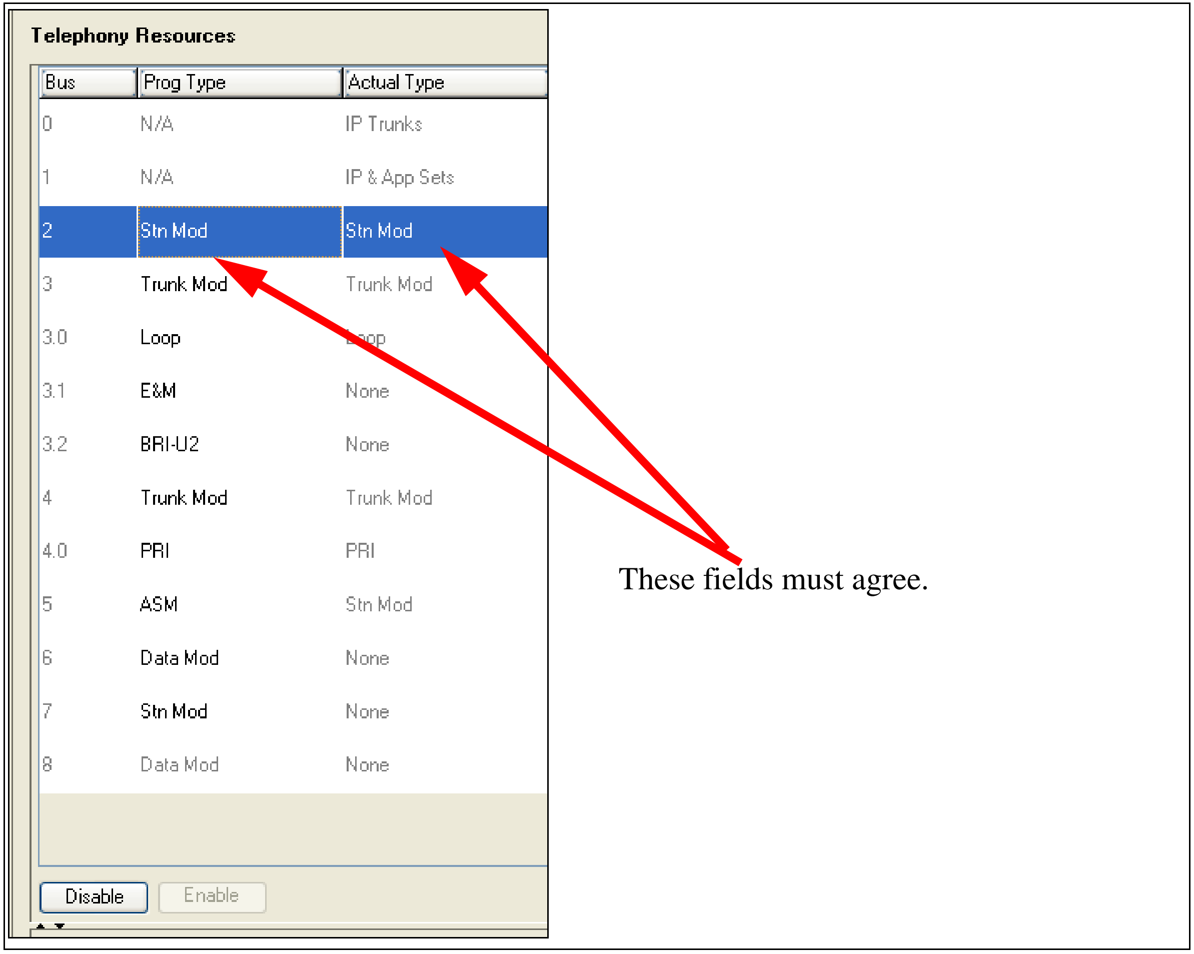
If these actions do not cause the fields to display correctly, you may have a damaged module or backplane. Try installing the module in a different media bay and retry the configuration. Refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) for information about removing and installing media bay modules.

4 The other headings on the Modules panel describe the current status of the modules, as described in the Table 19.

5 Your next step depends on which type of module you are configuring:

- If you are configuring a station or analog station module, ensure that the bus type is correct and the Programmed Bus Type field displays the correct module type. The **State** field displays **Enabled**, indicating that the module is active and ready to have telephones connected. Refer to "Module types and capacities" on page 104.
- If you are configuring trunk modules, you must now ensure each module associated with the DS30 bus is set up. This process is described in, "Defining trunk module types and settings" on page 106.

Figure 24 Confirming the Programmed Bus Type



Telephony Resources

Bus	Prog Type	Actual Type
0	N/A	IP Trunks
1	N/A	IP & App Sets
2	Stn Mod	Stn Mod
3	Trunk Mod	Trunk Mod
3.0	Loop	Loop
3.1	E&M	None
3.2	BRI-U2	None
4	Trunk Mod	Trunk Mod
4.0	PRI	PRI
5	ASM	Stn Mod
6	Data Mod	None
7	Stn Mod	None
8	Data Mod	None

These fields must agree.

Table 19 Bus record settings

Heading	Value	Description
Station module		
Number of sets	<digit>	This setting indicates the number of sets that are currently attached to the module.
Number of busy sets	<digit>	This setting indicates the number of sets that are currently using the module.
State	Enabled Disabled	This setting indicates the state of the module. Use the **Configuration** menu item to change this setting.
Trunk module, Analog Station Module or Data Module		
Number of busy ports	<digit>	This setting indicates how many ports on the module are currently being used.
State	Enabled Disabled	This setting indicates the current state of the module. Use the Configuration menu item to change this setting.

Module types and capacities

Refer to the following table for a description of the Bus types settings.

Table 20 Programmed Bus Types (Sheet 1 of 3)

Programmed Bus Type	Hardware unit	Capacity	Available line types (some line types are region-dependent)
Station module	• Digital Station Media Bay Module (DSM 16/16+ or DSM 32/32+)	Single density • DSM16/16+ = 1 per bus/16 digital sets per module • DSM 32/32+ = 2 buses/ 32 digital sets per module Double density • DSM16+ = 2 per bus/ 16 digital sets per module • DSM 32+ = 1 per bus/ 32 digital sets per module	N/A
	• 4X16 Media Bay Module (4X16) (counts as one DSM 16)	• 4X16 = 1 offset (trunk) and additional bus/ 16 digital sets	
	• Norstar station module (SM) connected to a FEM	• SM = 1 bus/16 digital sets	

Table 20 Programmed Bus Types (Sheet 2 of 3)

Programmed Bus Type	Hardware unit	Capacity	Available line types (some line types are region-dependent)
Analog station module	• Analog Station Media Bay Module (ASM 8)	Single density • ASM8 = 2 per bus/8 analog sets for each module Double density • ASM8 = 4 per bus/16 analog sets for each module	N/A
	• (Global) Analog Station Media Bay Module (GASM)	Single density • 2 per bus/8 analog sets for each module Double density • 4 per bus/8 analog sets for each module	Provides CLID passthrough, Message Waiting Indication* and Disconnect Supervision* (*available features depends on market profile)
	• Norstar analog station module (ASM) connected to a FEM	• FEM = 1 per bus/16 digital sets	N/A
Trunk module	• Digital Trunk Media Bay Module (DTM)	• DTM = 1 per bus/16 lines (max. three DTMs on a system)	• DTMs can be set to module types: Loop, E&M, DID, T1, PRI (NI-2 or ETSI are region-specific)
	• CLID Trunk Media Bay Module (CTM4 or CTM8) • Global Analog Trunk Module (GATM4 or GATM8) (released with BCM version 3.5)	• CTM4/GATM4=1 per offset/ 4 lines per module • CTM8/GATM8= 2 per bus/ 8 lines per module	• CTMs/GATMs can be set to module types: Loop
	• 4X16 Media Bay Module (4X16) (counts as one CTM)	• 4X16= 1 per offset (4 lines) and additional bus (station)	• 4X16s can be set to module type: Loop
	• BRI Media Bay Module (BRI)	• BRI= 3 per bus, 4 loops (8 lines) per module	• BRI can be set to module types BRI S/T, BRI U2, BRI U4 (setting must match physical module type). U2 and U4 are region-specific
Specialized modules	• Norstar trunk expansion modules, with Analog Trunk cards, connected to a FEM	• FEM= 1 per bus, can support up to three analog trunk cards (in one trunk expansion unit)/4 lines each	Norstar analog trunk cards: Loop, E&M, DID
	• DDI MUX	• 1 per bus	• T1 (telephony and data). Refer to "Configuring the DDI Mux module" on page 519.
	• Norstar trunk expansion modules with BRI cards, connected to a FEM	• FEM= 1 per bus, can support up to three BRI cards (in one trunk expansion unit)/4 loops each	• Norstar BRI cards: BRI S/T, BRI U2, BRI U4 (setting must match physical module type). U2 and U4 are region-specific
Data module	Refer to the data section of this book ("Configuring a data module" on page 526) for details about setting up a data module on DS30 08. This process includes any Norstar Data Modules connected to a FEM.		

Table 20 Programmed Bus Types (Sheet 3 of 3)

Programmed Bus Type	Hardware unit	Capacity	Available line types (some line types are region-dependent)
Legacy equipment	• DECT Media Bay Module (DECT)	• DECT= 1 per bus, 4 loops, supports 36 handsets	DECT (region-specific)

GATM (Global Analog Trunk module)

These trunk modules can be adjusted to have static trunk parameters (pre-BCM 3.5 software), or to allow the system to download new parameters when they become available (BCM 3.5 and later software). If the trunk is set to the latter state, trunk parameters are downloaded every time the module boots up, or when the parameters change while the module is working. You can also set line and telephone impedance to either 600ohm or 900ohm for modules with downloadable parameters. To set impedance, select **Configuration > Telephony > Lines > Active Physical Lines > Properties tab**. Refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) for details about the module.

Ports on Bus

Both types of modules have a **Ports on Module** table in the details panel. This panel shows the state of the port that connects the module to the trunk line or system device.

- Trunk modules - each port maps to an incoming line. Trunk port status is either equipped or unequipped.
- Station modules - each port maps to a connection to a system device (telephones, fax machines, doorphone). For station modules, port status is either equipped (device connected) or unequipped (no device connected). You can also determine what type of device is attached to the port.

Bus 01 and 08 note: Bus 01 and 08 ports on bus are virtual ports, since they connect to services supported by the MSC. These ports are used for such devices and services as IP telephones and voice mail traffic.

Defining trunk module types and settings

MSC bus numbers assigned to a trunk module display module numbers under the Bus column that correspond to the offset number set on the module DIP switches. For instance any trunk module that has an offset of 0, appears in the Module 1 column, and so on to a maximum of four modules (CTM4s/GATM4s).

The Module menu, that appears only under a Bus record that is configured for trunk modules, allows you to configure line or loop provisioning for the module associated with a particular bus. This record shows the number of lines or loops assigned to the module. It also provides the first and last loop or line number. These settings are read-only.

The fields that appear on the module screen vary depending on the module type you specify. Trunk lines can also require configuration of settings such as the protocol type/version, frame structure, clocking, and timers. These parameters depend on how the service provider interface that connects the module is configured.

Refer to the following topics:

- "Configuring the trunk module to line type" on page 107
- "Determining clock sources for DTMs or BRIs" on page 108
- "T1 interface parameters (region-specific)" on page 109
- "E1 parameters (region-specific)" on page 111
- "PRI Call-by-Call service selection" on page 112
- "Provisioning lines (PRI, T1, DASS2)" on page 114

Configuring the trunk module to line type

Follow these steps to define the modules to the system:

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the **Module** you want to program.
   The details panel for that bus appears. Figure 25 shows an example of the fields that display for a PRI module type.

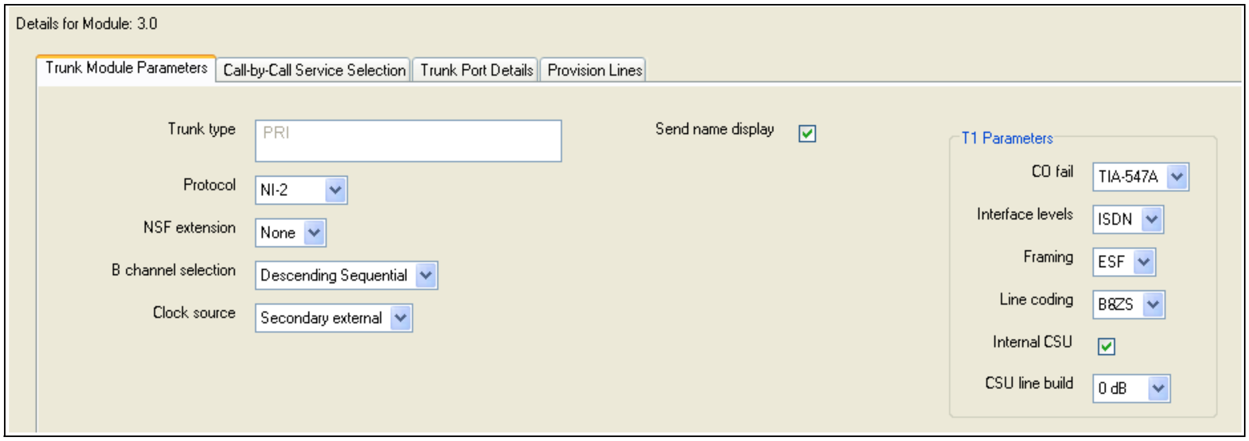
   The module installed in the system and Module Type must match the defined type of loop or trunk and associated services provided by the central office line that you intend to connect to the module.

   Note: When you configure a media bay module for PRI or BRI, the system may download new software to the module. This takes a couple of minutes to complete. Allow the download to finish before continuing to program the module.

   Tip: To refresh the record, you may need to click another navigation tree heading and then re-enter the module record you were working on.

   Check the settings to ensure they reflect the line requirements. Note that only some of the fields appear for all module types.
3. If your module is set to T1, PRI, or DASS2, refer to "To configure Call-by-Call services and the PRI lines" on page 175 to continue with the configuration.
4. After you complete your module configurations, refer to "Configuring lines" on page 147 to set up the lines the trunk modules will use.

Figure 25 Example of PRI module settings



Details for Module: 3.0

Trunk type: PRI

Protocol: NI-2

NSF extension: None

B channel selection: Descending Sequential

Clock source: Secondary external

Send name display ☑

T1 Parameters

CO fail: TIA-547A

Interface levels: ISDN

Framing: ESF

Line coding: B8ZS

Internal CSU ☑

CSU line build: 0 dB

Determining clock sources for DTMs or BRIs

Use the clock source list to designate the DTM or BRI on the system that obtains the timing reference for synchronization from the network.

Systems with digital interfaces need to synchronize to the network in order to function. Synchronization follows a hierarchical path. Each device (switch) obtains the network clock from the device above it in the synchronization hierarchy. The device then passes the network clock to the device below it in the synchronization hierarchy. The synchronization levels are referred to as strata.

BCM systems are stratum 4E equipment and are usually used as termination points in a network.

For each DTM and BRI, choose one of the following settings: **Primary external**, **Secondary external**, or **Internal**:

- Primary external — The DTM/BRI obtains the timing reference from the network and the system synchronizes to it. This is the default value for the ***first*** DTM in a BCM. Note that there should only be one defined Primary clock source on a System.
  Private network: If this system is in a private network and is intended to provide the master clock for that private network, the system must have one, and only one, Primary clock reference on a DTM or BRI. If this system is intended to act as clock master in a private network, then all clock sources should be set to Timing Master on this system.

- Secondary external — The DTM/BRI acts as a standby reference. If there are excessive errors on the Primary reference link, or the DTM/BRI designated as Primary reference fails, the Secondary DTM/BRI obtains the timing reference from the network to be used for system synchronization. This is the default value for the ***second*** DTM in a BCM.
  Private network: If this system is in a private network and is intended to provide the Master clock for that private network, then there should be no Secondary reference defined on any DTM/BRI. Note that there should only be one defined Secondary clock source on a system.

- Internal —The DTM/BRI does not obtain timing from the network, but transmits the internally-generated system timing, which is derived from the Primary/Secondary source, to equipment to which it is connected.
  Note that while in the absence of a DTM Primary clocking source a BRI module can be used for the primary timing reference, it is always recommended that, when possible, DTM(s) be used as primary (and secondary) clock sources and that any remaining DTMs/BRIs be set to Timing Master.

⚠ **Warning:** Changing the clock source may disconnect calls.
If you change the clock source for your system, you may cause your system DTM interface(s) to reset, resulting in dropped calls. Choose a suitable time to change the clock source and use the Page feature to inform users of possible service disruptions.

Timing within networks

In most T1/E1 network configurations, you need one DTM or BRI configured as Primary to act as a primary reference and obtain clocking from the network.

The only application where you might not have a DTM/BRI designated as a primary reference is in a private DTM/BRI network where your BCM system is connected to other equipment using T1/E1/BRI interface(s) that require a clock source and your system is designated as the clocking source for that private network.

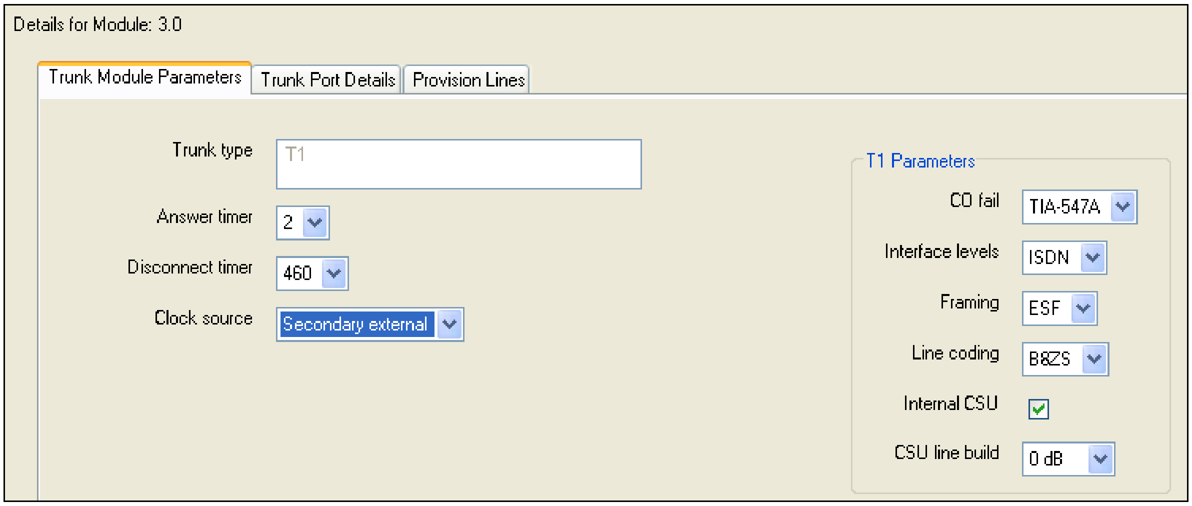
- If the other switches are to be clocked to your BCM system, *all* your DTMs/BRIs should be designated as Timing Master.
- If your BCM system has two DTMs, you cannot assign both DTMs as primary reference or both DTMs as secondary reference.
  You can have one Primary reference and one Secondary reference per system.
- A T1, PRI(T1), PRI(E1), or BRI can act as the clock source.

T1 interface parameters (region-specific)

The **T1 Parameters** details appear for module types that have been configured as T1 or PRI. It allows you to define a number of settings that are dependent on your T1 service provider settings.

1. Click **Configuration > Resources > Telephony Resources**.
2. Click a **T1** bus.
   The T1 details panel appears. See Figure 26.
3. Configure the T1 parameters. Refer to the information in Table 21.

Figure 26 T1 parameters



Details for Module: 3.0

Trunk type: T1

Answer timer: 2

Disconnect timer: 460

Clock source: Secondary external

T1 Parameters

CO fail: TIA-547A

Interface levels: ISDN

Framing: ESF

Line coding: B8ZS

Internal CSU: ☑

CSU line build: 0 dB

Table 21 T1 parameters

Attribute	Value	Description
CO fail	TIA-547A or TR62411	Select the carrier failure standard used by your T1 or PRI service provider. Consult your T1 or PRI service provider for the proper setting.
Interface levels	ISDN or PSTN	Define a loss plan setting. For more information, see "Interface levels" on page 110.
Framing	ESF or SF	Select the framing format used by your T1 or PRI service provider: Extended Superframe (ESF) or Superframe (SF). Contact your T1 or PRI service provider for the proper setting. (SF or Superframe is sometimes known as D4.)
Line coding	B8ZS or AMI	Define the encoding signals on a T1 line. Select the standard used by your T1 service provider. Contact your T1 service provider for the proper setting.
Internal CSU	On or Off	Turn the internal T1 channel service unit (CSU) on or off. For more information, see "Internal CSU" on page 111.
CSU line build	0, 7.5, or 15 dB	Set the gain level of the transmitted signal. This setting appears only when the Internal CSU is set to On.

Interface levels

The default Interface levels are the ISDN loss plan settings.

Check with your telecommunications service provider to determine if your BCM system is connected to a central office (CO) with digital network loss treatment (ISDN I/F levels) or analog network loss treatment (PSTN I/F levels).

The ISDN setting requires digital access lines (DAL) that have digital network loss treatment. On a DAL network, the PBX system administers the dB loss not than the CO. DALs may have ISDN signaling or digital signaling (for example, T1). The loss plan follows the Draft TIA-464-C loss plan, which uses a send loudness rating (SLR) of 8 dB. You must contact your service provider to get DAL network loss treatment on a line with digital signaling.

The PSTN setting requires analog access lines (AAL) that have analog network loss treatment and digital signaling. On an AAL(D) network, the CO administers the dB loss.

The loss plan follows the Draft TIA-464-C loss plan. The ISDN loss plan uses a send loudness rating (SLR) of 8 dB and a receive loudness rating (RLR) of 2 dB. The PSTN loss plan uses an SLR of 11 dB and an RLR of -3 dB. If you choose the wrong setting, the voice signal can be too loud or too soft.

Internal CSU

Internal CSU allows you to turn the internal T1 channel service unit on or off. The channel service unit gathers performance statistics for your T1 lines or PRI with public interface. Contact your service provider for the correct settings.

Note: You must disable the DTM before you can change this setting. See "Disabling/enabling a single module" on page 120 for details.

You can view the performance statistics for your T1 lines in Maintenance under the CSU stats heading. Before you set the internal CSU to off, you must ensure there is an external CSU connected to your T1 lines.

E1 parameters (region-specific)

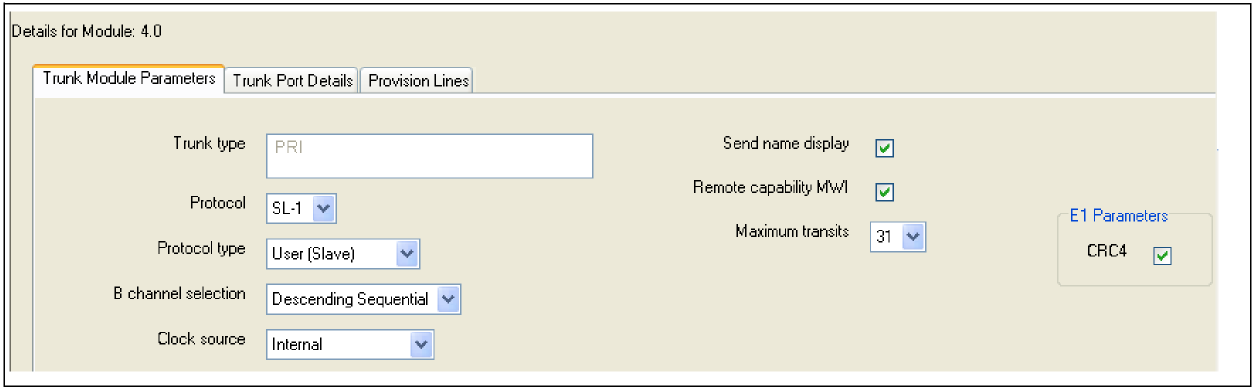
The E1 Parameters command appears for modules that have been configured as PRI in an E1 region. The CRC4 setting is the only selection in the E1 Parameters menu. CRC4 checking is enabled at the other end.

CRC (Cyclic Redundancy Checking) is a way for the CO to detect errors on the E1 link, not all CO providers use it.

The BCM must use the same setting as the CO.

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the **E1** bus.
   The details panel for the E1 bus appear.
3. Enable or disable the CRC4 parameter.

Figure 27 E1 Parameters



Details for Module: 4.0

Trunk type: PRI

Protocol: SL-1

Protocol type: User (Slave)

B channel selection: Descending Sequential

Clock source: Internal

Send name display ☑

Remote capability MWI ☑

Maximum transits: 31

E1 Parameters

CRC4 ☑

PRI Call-by-Call service selection

The following provides information about how to configure the PRI Call-by-call Service Selection, which is region-specific to North America, for a DTM set to a PRI Module type.

By default, incoming calls on a PRI are routed based on the Called Party Number information within the call request. The last number of digits of the called party number which matches the Received Number Length setting, are used as Receive Digits to find a target line.

For example, assume an incoming called party number is 800-555-1234. The received digit number length is 4, and the result is 1234. These last four digits are used to route the call.

In North American PRI, the Call-by-Call services allows alternate routing maps to be defined in various ways, depending on the protocol defined for this PRI.

Use this procedure to define call-by-call services:

1 Click **Configuration > Resources > Telephony Resources**.

2 Click a **PRI** bus.

3 Click on **Call-by-Call Service Selection**.
Table 22 lists the applicable services for the protocol defined on the Module <number> record.

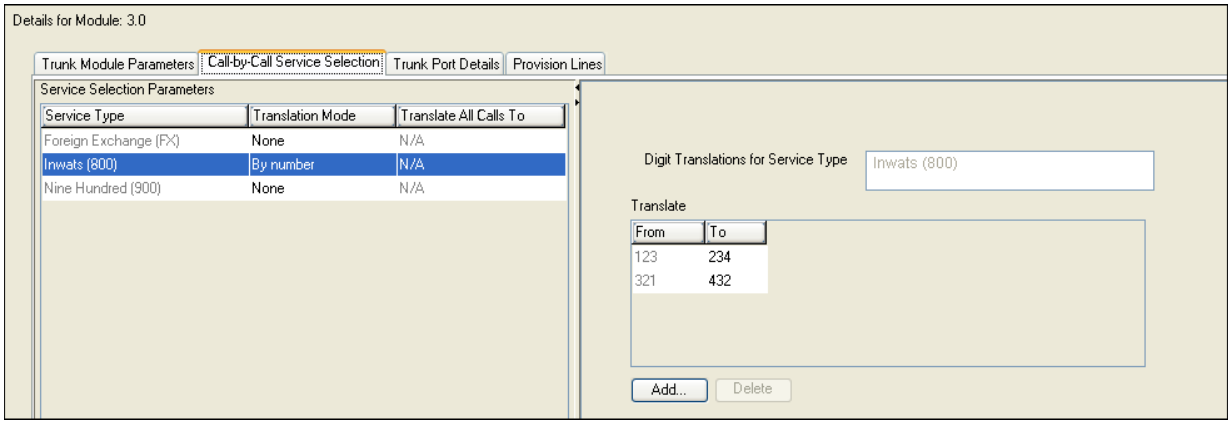
Table 22 Services available for each PRI protocol

	Services Available				
Protocol	**Foreign Exchg**	**Inwats (800)**	**Intl-800**	**Switched Digital (SDS)**	**Nine Hundred (900)**
NI-2	SID or All	By number or All	N/A	N/A	N/A
DMS100	SID or All	SID, By number, or All	N/A	N/A	N/A
DMS250	SID or All	SID, By number, or All	N/A	N/A	SID, or By number, or All
4ESS	N/A	By number or All	By number or All	By number or All	By number or All
SL-1	N/A	N/A	N/A	N/A	N/A

4 Select the service you want to change.
A configuration panel appears in the right frame. See Figure 28.

5 Table 23 shows the possible settings for the services.

Figure 28 Translation Mode



Details for Module: 3.0

Service Selection Parameters

Service Type	Translation Mode	Translate All Calls To
Foreign Exchange (FX)	None	N/A
Inwats (800)	By number	N/A
Nine Hundred (900)	None	N/A

Digit Translations for Service Type: Inwats (800)

Translate

From	To
123	234
321	432

Table 23 Module record values

Attribute	Value	Description
Translation Mode	None All By SID By Number	Define how the system maps incoming digits for this service type to the line number within the system. In all cases, the received digits are used to find a target line or to activate remote access. Default: None
Translation mode value definitions:		**None:** No mapping is applied. The last digits of the Called Party Number which match the Received Number Length setting are used as received digits. Note that if there is no called party number (may occur with some FX calls) the call will ring at the incoming trunk prime set.
		All: Allows you to define the received digits used for all calls with this service type, regardless of the called party number or service identifier (SID). For this option, all calls with this service type on this PRI will ring the same target line. Depending on the service type and the protocol, you may be able to map the called party number (By number) and the service identifier (SID).
		By SID: Allows you to associate different received digits with different calls of this service type based on the service identifier. **By Number:** Allows you to associate different received digits with different calls of this service type based on party number.
		Note: Any calls that do not match any entry defined in the map table will ring at the prime set.
Map Table	Select **Add** on the Map Table screen to create a new map entry From *digits* To *digits*	Enter the incoming line number to the internal line number, such as the target line.

Provisioning lines (PRI, T1, DASS2)

Use the **Provision lines** heading to provision and deprovision lines associated with a T1 PRI, E1 (DASS2), or BRI ST/U interface.

Provisioning a line or loop makes the line or loop available for system use. A deprovisioned line or loop is not available for use. If you are purchasing a partial PRI trunk, the lines that have not been assigned must be set to Deprovisioned.

The line number listed in each **Line** entry corresponds to the line numbers listed under the **Configuration > Telephony > Lines > Active Physical Lines** panel.

Provisioning a line

Note: All PRI lines are provisioned by default.

To provision a line

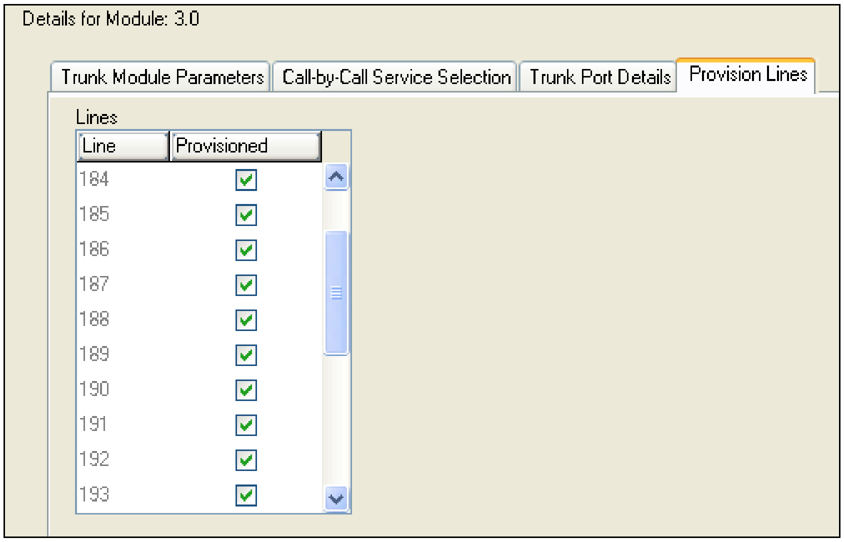
1 Click **Configuration > Resources > Telephony Resources**.

2 Click a bus (Bus 02 to 07) associated with the trunk module you want to provision.
The details panel for that module appears

3 Click the **Provision Lines** tab.
All the available lines appear.

4 Select the check box next to the line you want to provision.

Figure 29 Provision lines



Details for Module: 3.0

Lines

Line	Provisioned
184	☑
185	☑
186	☑
187	☑
188	☑
189	☑
190	☑
191	☑
192	☑
193	☑

Provisioning BRI loops/lines

To provision lines on a BRI module, you must first provision the loop on which the lines exist.

To provision BRI loops/lines

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the BRI module on which you want to provision loops.
   The details panels for that module appears
3. Select the **Provision Loops** tab.
   All the available loops appear.
4. Select the check box next to the loop you want to provision.
   The lines available to provision appear.
5. Select the check box next to the lines to provision.

Deprovisioning a line/loop

When you are not using a line/loop, or when you want to cancel it, you can deprovision that line or loop.

To deprovision a line or loop

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the module, which contains the lines or loops you want to deprovision.
   The details panel for that module appears.
3. Click the **Provision Lines** or **Provision Loops** tab.
   A list of the lines or loops assigned to the module appears.
4. Clear the **Provisioned** check box beside the line or loop you want to deprovision.
   The line or loop is deprovisioned.

Internally-driven channels

You cannot change headings for buses that are used for internal processing. This section describes how these buses fit into the system.

Bus 01 and Bus 08 provide access to telephony operations for internal processing, applications, and IP sets on the Business Communications Manager system. These two buses are commonly referred to as virtual buses since they have no external physical connections.

Bus 01 has 32 virtual ports. Bus 08 has 28 virtual ports. Each of these ports has one media channel associated with it. When IP telephones are assigned to the system, they will appear on these ports. The ports are allocated sequentially as telephones are added.

Bus 08 can also be used for a virtual data module (NA only) when a Business Communications Manager data service such as WAN service is activated. Refer to "To configure the DDI Mux" on page 522 for further programming. Note that Bus 08 does not display when there is a WAN active on the system.

By default, Bus 07 is used for a media bay module connection (2/6 channel split). However, if your system was set to a 3/5 DS30 split, then Bus 07 becomes a virtual bus with 32 ports. The headings under Bus 07 become invalid in this configuration. Refer to the "Configuring application resources" on page 81 for further details.

Working with the modules

When you need to find out information about a module, you can determine the status of any of the settings under the media bay modules headings. To correct a problem, or change a module setting, you may need to enable or disable a port, a module, or an entire bus. Refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) for more information on configuring Media Bay Modules.

Chapter 7
Managing modules

When you need to find out information about a module, you can determine the status of any of the settings under the media bay module headings. To correct a problem, or change a module setting, you may need to enable or disable a bus/module or select ports on the module. This section provides the procedures that describe:

- Disabling or enabling a bus or module
- Disabling or enabling a port channel setting

Disabling or enabling a bus or module

The following procedure describes the process for enabling or disabling a bus. This means that if there is more than one module assigned to the DS30 bus, all modules will be disabled.

To enable or disable a bus

1. Click **Configuration > Resources > Telephony Resources > Modules panel**, and then click the module you wish to enable/disable.
2. Click either the **Enable** or **Disable** button.
   The system prompts you to confirm your request.
3. Click **OK**.

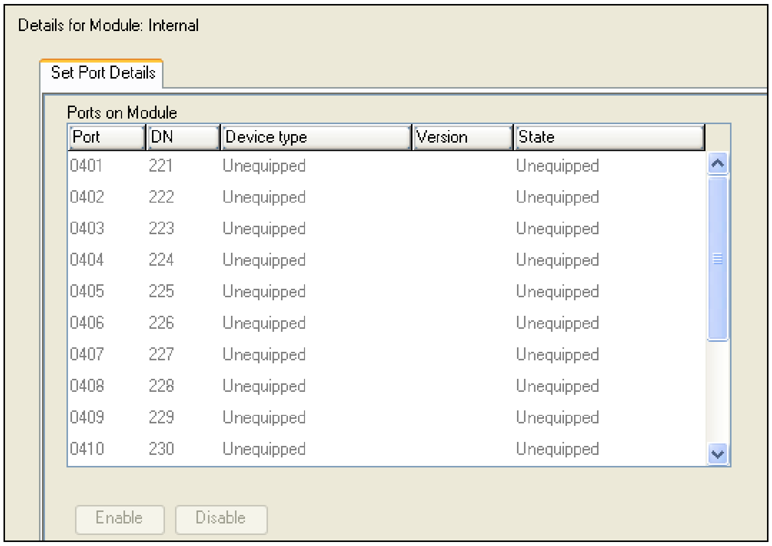
Disabling or enabling a port channel setting

If you need to isolate a problem or block access from the module, you may need to turn off individual port channels, rather than the entire module.

To turn a port channel on or off

1. On the **Configuration > Resources > Telephony Resources > Modules panel**, click the module supporting the port you want to enable/disable.
2. Select the port you want to enable/disable in the **Set Port Details** tab.
3. Click either the **Enable** or **Disable** button.
   The **State** field indicates the mode of operation for the port, as shown in Figure 30. If the port is enabled, this field shows unequipped unless a device is physically connected.

Figure 30 Set Port Details



Details for Module: Internal

Ports on Module

Port	DN	Device type	Version	State
0401	221	Unequipped		Unequipped
0402	222	Unequipped		Unequipped
0403	223	Unequipped		Unequipped
0404	224	Unequipped		Unequipped
0405	225	Unequipped		Unequipped
0406	226	Unequipped		Unequipped
0407	227	Unequipped		Unequipped
0408	228	Unequipped		Unequipped
0409	229	Unequipped		Unequipped
0410	230	Unequipped		Unequipped

Note: A trunk media bay module has no changeable settings on the Trunk Port Details record.

Chapter 8
Configuring telephony resources

The Telephony Resources panel allows you to view and configure the information for the modules that support the digital/analog/ISDN lines for the system and the gateways that support the Voice over IP (VoIP) trunks. This provides a cohesive view of your telephony communications channels for the system.

The following paths indicate where to configure telephony resources in Element Manager and through Telset Administration:

- Element Manager: **Configuration** > **Resources > Telephony Resource**s
- Telset interface: ****CONFIG > Hardware** (does not allow you to configure VoIP trunks or IP telephones)

The following table provides links to descriptions of each subpanel.

Panel	Tasks
“Telephony Resources table” on page 120	“Managing modules” on page 117
“Media bay module panels” on page 122	“Configuring the trunk module to line type” on page 107
“Trunk Module Parameters” on page 122	
“Port details” on page 128	
“Call-by-Call Service Selection” on page 126	
Also refer to:	“Dialing plan: Private network settings” on page 317
	“Call security: Remote access packages” on page 463
“Provisioning module lines/loops” on page 130	
“IP telephones” on page 131	
“IP Terminal Global Settings” on page 131	“Registering Nortel 20XX and 11XX IP telephones” in the *BCM 4.0 Telephony Device Installation Guide* (N0060609)
“IP telephone set details” on page 132	
“Voice over IP trunks” on page 133	
	“Configuring VoIP trunk gateways” on page 409
	“VoIP interoperability: Gatekeeper configuration” on page 417
“Routing table” on page 134	
“IP Trunk Settings” on page 136	
“H323 Settings” on page 137	
“H323 Media Parameters” on page 140	“Setting up VoIP trunks for fallback” on page 423
“SIP Settings” on page 142	
“SIP Media Parameters” on page 144	

Panel	Tasks
“SIP URI Map” on page 146	
Click the navigation tree heading to access general information about user management.	

The top frame of this panel displays a table showing each type of module and the VoIP trunks that are assigned to the system, either through connections to a media bay module or by applying the required keycodes (VoIP trunks).

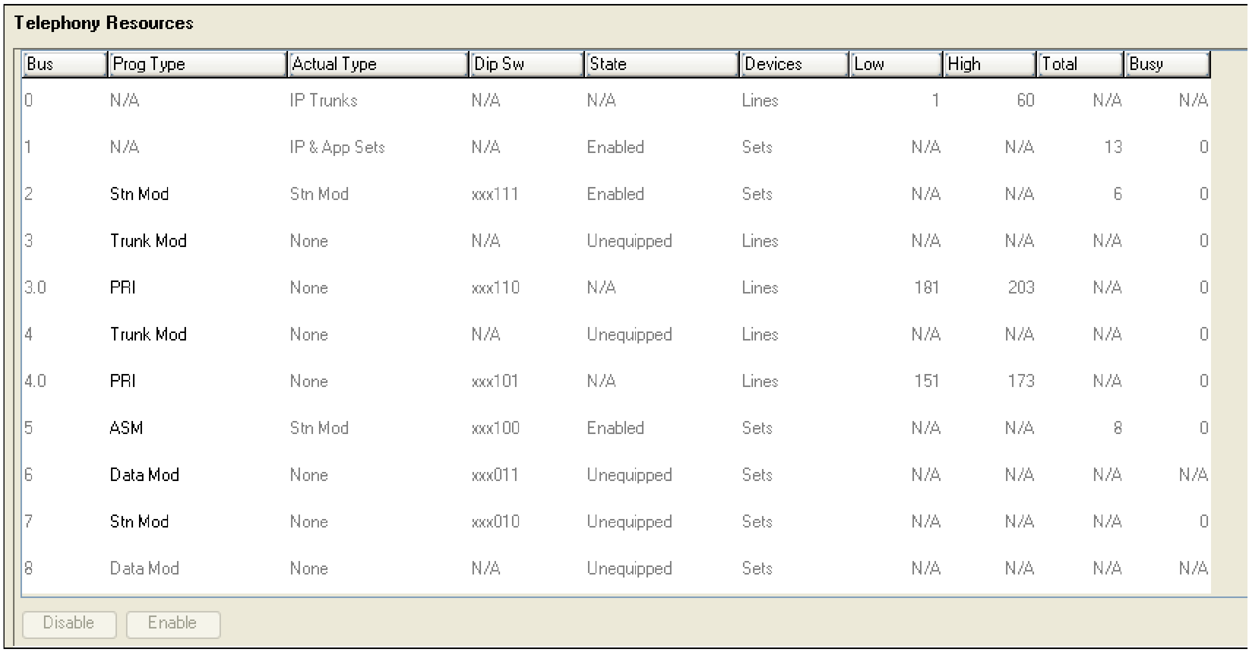
Selecting a table listing provides access to the special settings for each type of resource in tabbed panels that appear in the lower window.

Telephony Resources table

The top-level panel shows a list of active modules and VoIP gateways and IP telephone IP network information.

Click the line for the resource you want to view or configure.

Figure 31 Telephony Resources table



Telephony Resources

Bus	Prog Type	Actual Type	Dip Sw	State	Devices	Low	High	Total	Busy
0	N/A	IP Trunks	N/A	N/A	Lines	1	60	N/A	N/A
1	N/A	IP & App Sets	N/A	Enabled	Sets	N/A	N/A	13	0
2	Stn Mod	Stn Mod	xxx111	Enabled	Sets	N/A	N/A	6	0
3	Trunk Mod	None	N/A	Unequipped	Lines	N/A	N/A	N/A	0
3.0	PRI	None	xxx110	N/A	Lines	181	203	N/A	0
4	Trunk Mod	None	N/A	Unequipped	Lines	N/A	N/A	N/A	0
4.0	PRI	None	xxx101	N/A	Lines	151	173	N/A	0
5	ASM	Stn Mod	xxx100	Enabled	Sets	N/A	N/A	8	0
6	Data Mod	None	xxx011	Unequipped	Sets	N/A	N/A	N/A	N/A
7	Stn Mod	None	xxx010	Unequipped	Sets	N/A	N/A	N/A	0
8	Data Mod	None	N/A	Unequipped	Sets	N/A	N/A	N/A	N/A

The Telephony Resources table fields are described in Table 24.

Table 24 Telephony Resources table fields

Attribute	Value	Description
Bus	<read-only>	<read-only> 1-XX
Prog Type	<read-only> FEM ASM/ASM+ GATM4 DSM16 DSM32/ DSM32+ 4X16 Combo DTM-T1 DTM-PRI CTM4/GATM4 CTM8/GATM8 BRIM Empty	This field indicates the type of module assigned to each location. ASM/GASM: Analog and Global Analog Station Modules provide four connections for four analog telephones. GATM8: Global Analog Trunk Module with four trunk line connections. DSM16 or DSM32/DSM32+: Digital Station Module with 16 and 32 telephone connections, respectively. 4X16 Combo: A module with a four-trunk analog board and a 16-port digital station module. BRI/ST, BRI/U2, BRI/U4 DTM-T1 DTM-PRI Empty: No module is currently connected.
Actual Type		
Dip Sw	<read-only>	Confirm that the dip switch settings match the intended DS30 bus placement.
State	Enabled Disabled Unequipped	Indicates the state of the module or bus: Enabled: module is installed and working Disabled: module is installed but has been disabled or is down for another reason Unequipped: there is no module installed on this bus
Devices	Set Lines	Lists the type of device configured on the bus.
Low	<digits>	This field indicates the lowest setting for one of the following: The range of lines the module/VoIP supports The range of loops the module supports (BRI) The range of DNs the module/IP telephony supports.
High	<digits>	This field indicates the highest setting for one of the following: The range of lines the module/VoIP supports The range of loops the module supports (BRI) The range of DNs the module/IP telephony supports.
Total	<XX> Lines, loops or Sets	This field indicates the total number of lines, loops or DNs that the module supports.
Busy	1-X	This field indicates the current activity for the devices or lines attached to the module.

Media bay module panels

The panel tabs described in this section appear when a module table entry is selected on the Telephony Resources panel.

Note that the four trunks connected to the core module are also indicated in the table when they are active. These trunks are analog trunks.

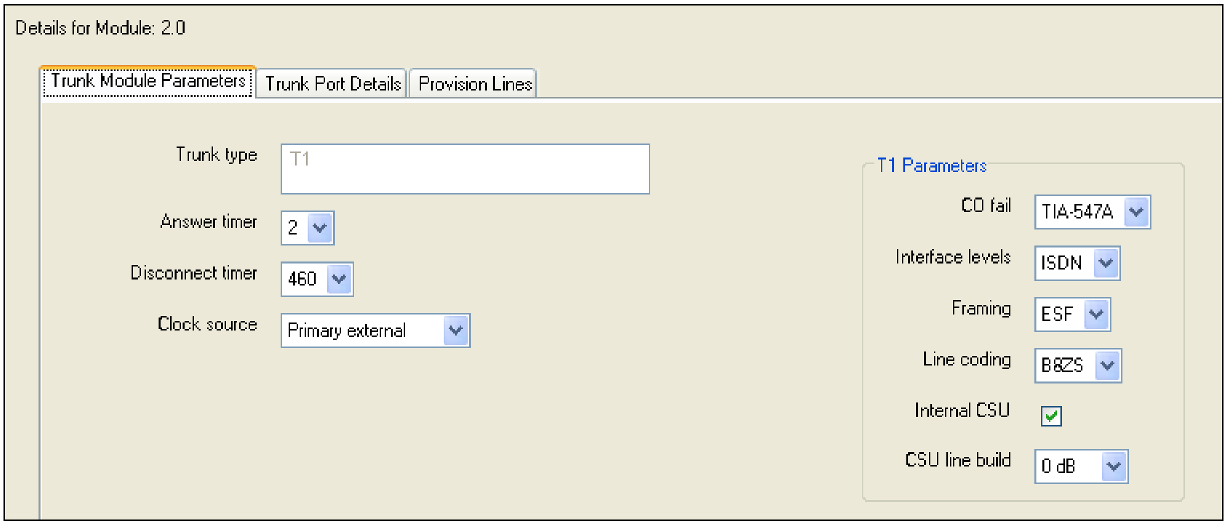
Refer to the following panel descriptions:

- "Trunk Module Parameters"
- "Port details" on page 128

Trunk Module Parameters

The Trunk Module Parameters tab shows the information that is unique to the type of trunk module selected in the main Modules list.

Figure 32 Trunk Module Parameters subpanel



Details for Module: 2.0

Trunk type: T1

Answer timer: 2

Disconnect timer: 460

Clock source: Primary external

T1 Parameters

CO fail: TIA-547A

Interface levels: ISDN

Framing: ESF

Line coding: B8ZS

Internal CSU: ☑

CSU line build: 0 dB

Table 25 describes the possible fields, trunk module parameters, and an indication of which types of modules use each setting.

Table 25 Module parameters values (Sheet 1 of 4)

Attribute	Value	Module/line type
Trunk type		**All trunks**
	Indicates the type of trunks. This field is read-only for all modules except DTM modules.	
Trunk mode	DS/CLID, Global, Legacy	**Loop**
	• DS/CLID: displays for old North American LS/DS or CLID analog trunk modules, the old analog MBM, or the GATM with North American DIP switch settings. • Global: displays for the GATM MBM with no regional DIP switches set. • Legacy: displays for all other (old) analog trunk modules	

Table 25 Module parameters values (Sheet 2 of 4)

Attribute	Value	Module/line type
Protocol	NI-2, DMS100, DMS250, AT&T4ESS, SL-1, Euro, ETSI Q.Sig	**PRI**
	Choose the trunk protocol used by your service provider. The supported protocols are: **PRI-T1:** NI-2, DMS100, DMS250, AT&T4ESS, SL-1 **PRI-E1:** ETSI QSIG, Euro, SL-1 **Note:** SL-1 and ETSI QSIG require an MCDN keycode to display. **BRI:** Protocol can also be selected on BRI T-loops under the **Configuration** > **Resources** > **Telephony Resources**. **Note:** Always check the line protocol with the central office.	
NSF Extension	None, WATS, ALL	**PRI**
	The Network Specific Facilities (NSF) information element is used to request a particular service from the network. Settings are based on the type of switch to which the line connects. Suggested settings: DMS100/250: NONE Siemens ESWD, Lucent 5ESS: WATS GTD5, DMS10: ALL When you select **NONE**, the NSF extension bit is not set for any service. When you select **WATS**, the NSF extension bit is set for unbanded OUTWATS calls. When you select **ALL**, the NSF extension is always set for all CbC services. Appears only for NI protocol.	
Protocol type	User, Network	**PRI**
	When you select SL-1 protocol, an additional setting, Protocol type, appears. SL-1 protocol is a private networking protocol. This allows you to designate a BCM node as a Network (controller). The default setting is User (client). In public network configurations, the CO is generally considered the Network side or controller. Applies to SL-1 protocol only.	
B-channel selection sequence	Ascending Sequential Descending Sequential	**PRI**
	Defines how B-channel resources are selected for call processing.	
Answer timer	1, 2, 3, 4, or 5 sec.	**E&M** **PRI**
	Set the minimum duration of an answer signal before a call is considered to be answered.	
Disconnect timer	60, 100, 260, 460, or 600 milliseconds	**Loop** **T1**
	Specify the duration of an Open Switch Interval (OSI) before a call on a supervised external line is considered disconnected. This setting must match the setting for the line at the central office (CO). You must enable disconnect supervision by changing the Line **Trunk mode** attribute. Under the Telephony Services sub-heading, choose Lines and Line/trunk Data.	

Table 25 Module parameters values (Sheet 3 of 4)

Attribute	Value	Module/line type
Clock Source	Primary External Secondary External Internal	T1 PRI *BRI S/T *BRI U2 *BRI U4 DASS2
	Designates whether the DTM/BRI acts as a primary or secondary timing component for an external timing source or as the internal timing source. **Note:** A BRI module can be programmed with primary/secondary clock source, however, it is recommended that a BRI module always be set to Internal if a DTM exists on the system to be the Primary External clock source. **Warning: Changing the clock source may disconnect calls**. If you change the clock source for your system, you may cause your system DTM interface(s) to reset, resulting in dropped calls. Choose a suitable time to change the clock source and use the Page feature to inform users of possible service disruptions.	
Send Name Display	Select or clear	PRI *BRI QSIG
	When you select this check box, the system sends a specified outgoing name display (OLI) from the calling telephone. Appears only for Protocols: SL-1, NI, DMS100, DMS250, or PRI QSIG.	
Remote Capability MWI	Select or clear	PRI
	This setting allows you to indicate MWI compatibility on the specific loop(s) that you are using to connect to the central voice mail system on a Meridian 1 which has the MWI package installed, with the RCAP setting set to MWI. Appears only for SL-1 protocol.	
Overlap receiving		BRI
	Supports target lines in markets which use Overlap receiving signalling on the BRI trunks. Overlap receiving must be configured for each BRI loop. After every digit is received at the ISDN layer, Target Lines are checked for matches. If a full match is made, the call is routed immediately to the target line without waiting for additional digits.	
Local Number Length		BRI
	When Overlap receiving is enabled on the trunks, this number determines how many incoming digits need to match the target line numbers to be considered a call for that target line.	
Host node	M1, Embark, IDPX, DSM	DNPSS
	DPNSS cards connected to Embark switches have a different way of handling call diversion, therefore, when you provision a DTM for DPNSS, you must indicate what type of switch the lines are connected to. When you select the Embark switch, calls are diverted using the Call Forwarding feature instead of call diversion.	
Local Number Length		DPNSS
	This number allows the system to determine how many digits to read on an incoming call to determine that the call is meant for this system.	
Maximum Transits	Default: 31	PRI
	Indicate the maximum number of times that a call will be transferred within the SL-1 network before the call is dropped. Protocol must be set to SL-1 to display this field.	

Table 25 Module parameters values (Sheet 4 of 4)

Attribute	Value	Module/line type
T1 parameters		
CO fail		T1 PRI
	Specify a carrier failure standard (T1A-5474, TR62411)	
Interface levels	ISDN, PSTN	T1 PRI
	Define a loss plan setting. For more information, see “Interface levels” on page 125.	
Framing	ESF, SF	T1 PRI
	Select the framing format used by your T1 or PRI service provider: Extended Superframe (ESF) or Superframe (SF). Contact your T1 or PRI service provider for the proper setting. (SF or Superframe is sometimes known as D4.)	
Line coding	B8ZS, AMI	T1 PRI
	Define the encoding signals on a T1 line. Select the standard used by your T1 service provider. Contact your T1 service provider for the proper setting.	
Internal CSU	<check box>	T1 PRI
	Turn the internal T1 channel service unit (CSU) on or off. For more information, see “Internal CSU” on page 126.	
CSU line build	0, 7.5, or 15 dB	T1 PRI
	Set the gain level of the transmitted signal. This setting appears only when the Internal CSU is Enabled.	
DSX1 build	000-100, 100-200, 200-300, 300-400, 400-500, 500-600, or 600-700 feet	T1 PRI
	Set the distance between BCM and an external channel service unit. This setting only appears when the Internal CSU is Disabled. Contact your service provider for the proper settings.	
CRC4	<check box>	E1 PRI
	Ensure this is enabled or disabled to match the service provider Cyclic Redundancy Check (CRC4) setting for the trunk.	

Station modules do not have any configurable module parameters.

Interface levels

The default Interface levels are the ISDN loss plan settings. Also refer to “ISDN overview” on page 701.

Check with your telecommunications service provider to determine if your BCM system is connected to a central office (CO) with digital network loss treatment (ISDN I/F levels) or analog network loss treatment (PSTN I/F levels).

The ISDN setting requires digital access lines (DAL) that have digital network loss treatment. On a DAL network, the PBX system administers the dB loss, not the CO. DALs may have ISDN signaling or digital signaling (for example, T1). The loss plan follows the Draft TIA-464-C loss plan, which uses a send loudness rating (SLR) of 8 dB. You must contact your service provider to get DAL network loss treatment on a line with digital signaling.

The PSTN setting requires analog access lines (AAL) that have analog network loss treatment and digital signaling. On an AAL(D) network, the CO administers the dB loss.

The loss plan follows the Draft TIA-464-C loss plan. The ISDN loss plan uses a send loudness rating (SLR) of 8 dB and a receive loudness rating (RLR) of 2 dB. The PSTN loss plan uses an SLR of 11 dB and an RLR of -3 dB. If you choose the wrong setting, the voice signal can be too loud or too soft.

Internal CSU

Internal CSU allows you to turn the internal T1 channel service unit on or off. The channel service unit gathers performance statistics for your T1 lines or PRI with public interface. Contact your service provider for the correct settings.

You can view the performance statistics for your T1 lines in Maintenance under the CSU stats heading. Before you set the internal CSU to off, you must ensure there is an external CSU connected to your T1 lines.

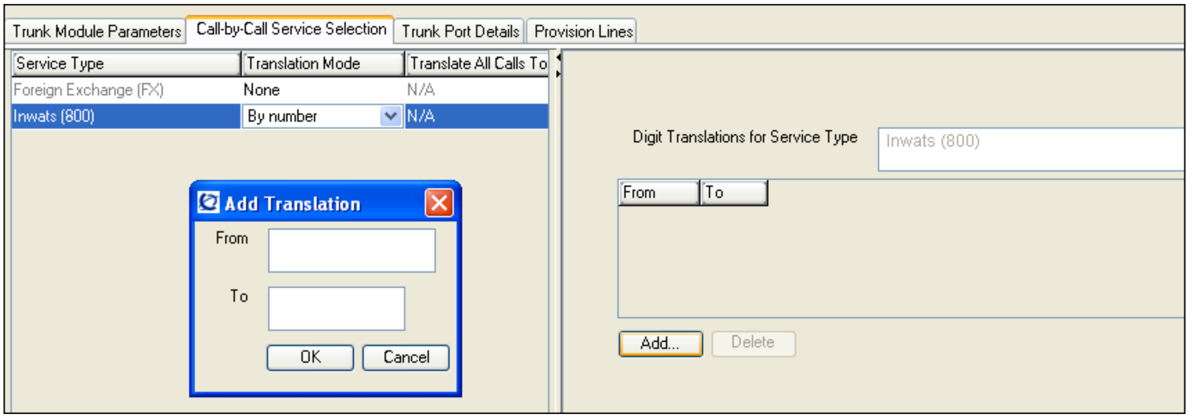
Call-by-Call Service Selection

This section provides information about how to configure the PRI Call-by-call Service Selection, which is region-specific to North America, for a DTM set to a PRI Module type.

By default, incoming calls on a PRI are routed based on the Called Party Number information within the call request. The last number of digits of the called party number that match the Received Number Length setting are used as Receive Digits to find a target line.

In North American PRI, the Call-by-Call services allows alternate routing maps to be defined in various ways, depending on the protocol defined for this PRI.

Figure 33 Call-by-Call Service Selection subpanel



Service Type	Translation Mode	Translate All Calls To
Foreign Exchange (FX)	None	N/A
Inwats (800)	By number	N/A

Digit Translations for Service Type: Inwats (800)

Table 26 describes the fields shown on the Call-by-Call Service Selection tab panel.

Table 26 Call-by-Call Service selection panel fields

Attribute	**Value**	**Description**
Service Type	Foreign Exchange Inwats (1-800) Intl-800 Digital (SDS) 900	Refer to "CbC services available by switch protocol" on page 128.
Translation Mode	None All By SID By Number	Define how the incoming digits get mapped to line numbers (target lines or DISA/AUTO DNs) within the system.
Translate All Calls To		Enter the appropriate information for the mode chosen.
Actions		
Add	1. On the Modules table, select the PRI module you want to configure. 2. Select the Service Type record to which you want to add Digit translations. 3. Under the Translate table, click **Add**. 4. Enter the appropriate information in the From and To fields on the dialog box. 5. Click **OK** on the dialog to save the translation range.	
Delete	1. On the Modules table, select the PRI module record you want to delete. 2. Select the Service Type record from which you want to delete Digit translations. 3. On the Translate table, select one or more ranges to delete. 4. Click **Delete**. 5. Click **OK** on the confirmation dialog to delete the digit translation range.	

CbC services available by switch protocol

Table 27 lists the applicable services for the protocol defined on the Module record.

Table 27 Services available for each PRI protocol

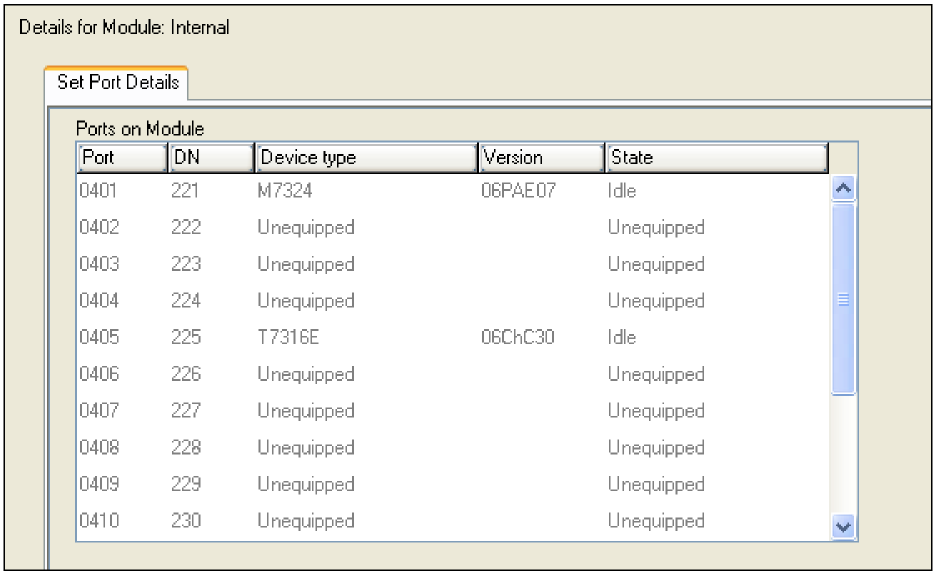
	Services Available				
Protocol	**Foreign Exchg**	**Inwats (800)**	**Intl-800**	**Switched Digital (SDS)**	**Nine Hundred (900)**
NI	SID or All	By number or All	N/A	N/A	N/A
DMS-100	SID or All	SID, By number, or All	N/A	N/A	N/A
DMS-250	SID or All	SID, By number, or All	N/A	N/A	SID, or By number, or All
4ESS	N/A	By number or All	By number or All	By number or All	By number or All

Port details

Both trunk and analog modules show port details. Ports settings are directly related to the physical ports into which the PSTN lines or telephony devices connect on the media bay modules.

The station module Port Details panel is illustrated in Figure 34. The trunk module Port Details panel is illustrated in Figure 35.

Figure 34 Station module Port Details panel

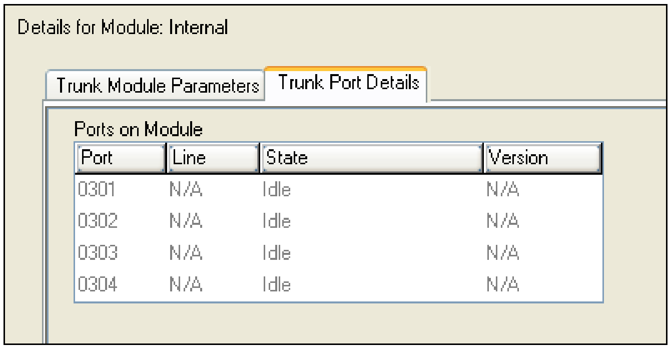

Details for Module: Internal

Set Port Details

Ports on Module

Port	DN	Device type	Version	State
0401	221	M7324	06PAE07	Idle
0402	222	Unequipped		Unequipped
0403	223	Unequipped		Unequipped
0404	224	Unequipped		Unequipped
0405	225	T7316E	06ChC30	Idle
0406	226	Unequipped		Unequipped
0407	227	Unequipped		Unequipped
0408	228	Unequipped		Unequipped
0409	229	Unequipped		Unequipped
0410	230	Unequipped		Unequipped

Figure 35 Trunk module Port Details panel


Details for Module: Internal

Trunk Module Parameters | Trunk Port Details

Ports on Module

Port	Line	State	Version
0301	N/A	Idle	N/A
0302	N/A	Idle	N/A
0303	N/A	Idle	N/A
0304	N/A	Idle	N/A

Table 28 describes the fields shown on the Port Values tab panel.

Table 28 Port Values tab (Sheet 1 of 2)

Attribute	**Value**	**Module type**
Port #	Read-only	**All modules**
	• These are the port numbers of the physical device.	
Device type	Read-only	**All modules**
	• This is the type of module.	
Line #	00X-XXX	**CTM/ GATM4** **CTM/ GATM8** **Combo** **DTM-T1** **DTM-PRI** **BRI-T**
	The number of lines depends on the module type.	
Call State or State	Idle Active Deprovisioned	**All modules**
	This field indicates whether a module line or DN is in use or even provisioned.	

Table 28 Port Values tab (Sheet 2 of 2)

Attribute	Value	Module type
Version	<read-only>	**All modules**
	This field indicates the version of firmware running on the module.	
DN	XXXX	**ASM/ GASM** **DSM**
	Each port supports one telephone, hence, one DN record.	
Addons		**All modules**
	Indicates auxiliary items added to the telephony devices or trunks	
	Add-on	This is a list number.
	Type	This field indicates the type of add-on, such as a KIM module.
	Version	This field indicates the version of firmware running on the add-on device.

Provisioning module lines/loops

There are three provisioning subpanels, which can be accessed in Element Manager at **Configuration > Resources > Telephony Resources**. The tabbed provisioning panel that appears depends on the type of module that is selected on the Telephony Resources table.

The provisioning subpanels are as follows:

- The Provision Lines tab panel is used for all trunks except DPNSS and BRI loops.
- The DPNSS module displays the Provision Virtual Channels tab panel.
- BRI loops require an extra step, so the Provision Loops tab panel appears when a BRI module is selected.

Table 29 describes the fields on these panels.

Table 29 Provisioning panels (Sheet 1 of 2)

Field	Value	Description
Provision Lines tab		
Line	<line number>	This is a list of the lines assigned to the module.
Provisioned	<check box>	If the check box is selected beside a line, that line is available for call traffic.
Provision Virtual Channels tab		
Virtual Channel	<read-only>	A virtual channel assigned to the DPNSS module.
Provisioned	<check box>	If the check box is selected beside a channel, that channel is available for call traffic.
Provision Loops tab		
Loop	<loop-number>	These are the loop numbers assigned to the selected BRI module. Modules have four loops, but only loops designated as T-loops require provisioning.

Table 29 Provisioning panels (Sheet 2 of 2)

Field	Value	Description
Provisioned	<check box>	If the check box is selected beside a loop, that loop has lines that can be provisioned.
Line	<line number>	Each loop as two lines assigned. You can provision or deprovision these lines individually.
Provisioned	<check box>	If the check box is selected beside a line, that line is available for call traffic.

IP telephones

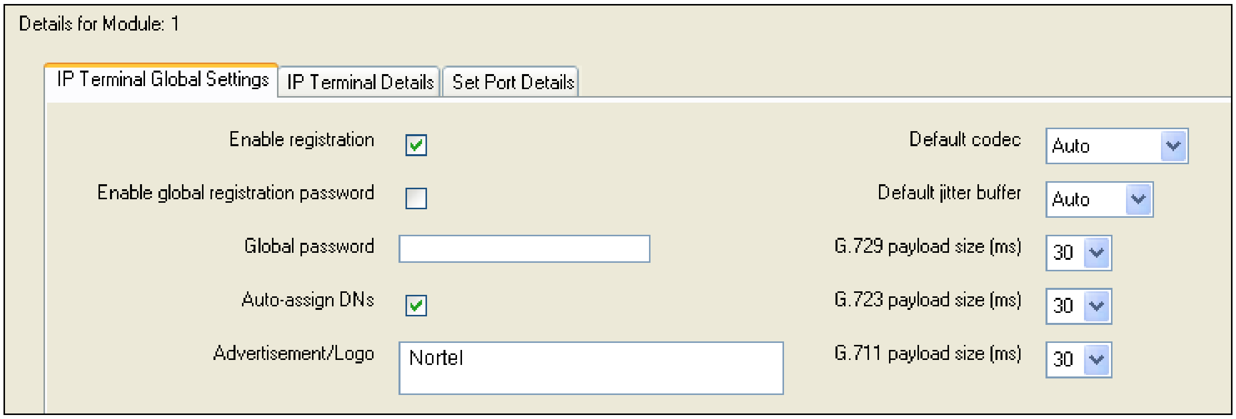
The tabbed panels described in the topics below appear when an IP terminals entry is selected on the Telephony Resources table.

- "IP Terminal Global Settings"
- "IP telephone set details" on page 132

IP Terminal Global Settings

The parameters on the IP Terminal Global Settings subpanel affect all Nortel 1120E/1140E/20XX IP telephones. This is also the panel you use to allow these telephones to register to the system, and to turn off registration once you have registered all the telephones.

Figure 36 IP Terminal Global Settings subpanel



Details for Module: 1

Enable registration ☑
Enable global registration password ☐
Global password
Auto-assign DNs ☑
Advertisement/Logo: Nortel
Default codec: Auto
Default jitter buffer: Auto
G.729 payload size (ms): 30
G.723 payload size (ms): 30
G.711 payload size (ms): 30

Table 30 defines the fields on this panel and indicates the lines.

Table 30 IP terminal Global panel fields (Sheet 1 of 2)

Field	Value	Description
Enable registration	<check box>	Select to allow new IP clients to register with the system. **Warning:** Remember clear this check box when you have finished registering the new telephones.
Enable global registration password	<check box>	If you want to require the installer to enter a password when IP telephones are configured and registered to the system, select this check box.

Table 30 IP terminal Global panel fields (Sheet 2 of 2)

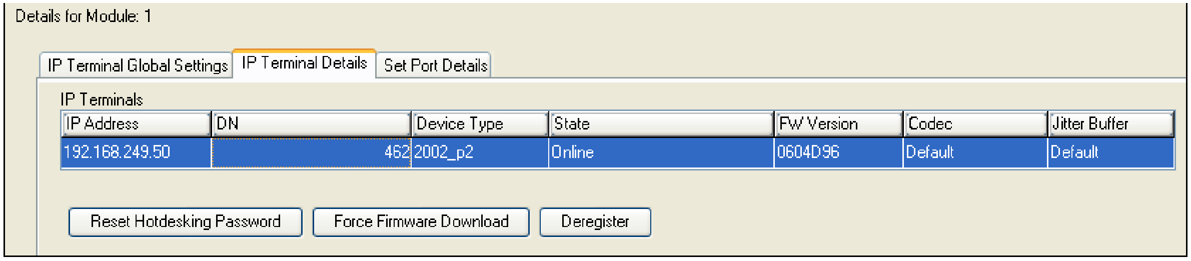
Field	**Value**	**Description**
Global password	<10 alphanumeric> Default: bcmi (2264)	If the Enable global registration password check box is selected, enter the password the installer will enter on the IP telephone to connect to the system. If this field is left blank, no password prompt occurs during registration.
Auto-assign DNs	<check box>	If selected, the system assigns an available DN as an IP terminal requests registration. It does not prompt the installer to enter a set DN. **Note:** For this feature to work, **Registration** must be selected and **Password** must be blank. If not selected, the installer receives a prompt to enter the assigned DN during the programming session. **Note:** Refer to the Caution notice at the top of this section.
Advertisement/Logo	<alphanumeric string>	Any information in this field appears on the display of all IP telephones. For example, your company name or slogan.
Default codec	Auto G.711-aLaw G.711-uLaw G.723 G.729 G.729 + VAD G.723 + VAD	If the IP telephone has not been configured with a preferred codec, choose a specific codec that the IP telephone will use when it connects to the system. If you choose **Auto**, the IP telephone selects the codec. If you are unsure about applying a specific codec, ask your network administrator for guidance.
Default jitter buffer	None Auto Small Medium Large	Choose one of these settings to change the default jitter buffer size: None: Minimal latency, best for short-haul networks with good bandwidth. Auto: The system will dynamically adjust the size. Small: The system will adjust the buffer size, depending on CODEC type and number of frames per packet to introduce a 60-millisecond delay. Medium: 120-millisecond delay Large: 180-millisecond delay
G.729 payload size (ms)	10, 20, 30, 40, 50, 60 Default: 30	Set the maximum required payload size, per codec, for the IP telephone calls sent over H.323 trunks. **Note:** Payload size can also be set for Nortel IP trunks. Refer to "Configuring VoIP trunk media parameters" on page 410.
G.723 payload size (ms)	30	
G.711 payload size (ms)	10, 20, 30, 40, 50, 60 Default: 20	

IP telephone set details

Once a Nortel 1120/1140 or 20XX IP telephone registers with the system, this panel displays the terminal parameters.

The telephone is identified to the system by its IP address, so this cannot be changed. If you need to change the IP address of a telephone, you need to deregister the telephone and then register it again with the new IP address.

Figure 37 IP Terminal Details (Telephony Resources) subpanel



IP Address	DN	Device Type	State	FW Version	Codec	Jitter Buffer
192.168.249.50	462	2002_p2	Online	0604D96	Default	Default

Table 31 describes the fields on this panel.

Table 31 IP terminal fields

Field	Value	Description
IP Address	<read-only>	If the telephone is using DHCP or partial DHCP, this may vary.
DN	<DN>	This is the DN record that defines the system parameters for the telephone.
Device Type	<read-only>	This is the type of IP telephone.
State	<read-only>	Indicates if the device is online.
FW Version	<read-only>	Current version of telephone software.
Codec	Default G.711-aLaw G.711-uLaw G.711+VAD G.729 G.729+VAD G.723 Auto	Specifying a non-default Codec for a telephone allows you to override the general setting. You might, for example, want to specify a low bandwidth Codec (G.729) for a telephone that is on a remote or busy sub-net. **Note:** You can only change the codec on a configured IP telephone if it is online to the system, or if Keep DN Alive is enabled for an offline telephone.
Jitter Buffer	Auto Default None Small Medium Large	Increase the jitter buffer size for any telephone that has poor network connectivity to the system. **Note:** You can only change the jitter buffer on a configured IP telephone if it is online to the system, or if Keep DN Alive is enabled for an offline telephone.
Actions		
Reset Hotdesking Password	This button allows you to reset the hotdesking password for a telephone. Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).	
Force Firmware Download	This button downloads the firmware from the system to the selected telephone. Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).	
Deregister	This button allows you to deregister the selected telephone. Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).Refer to the *BCM 4.0 Device Configuration Guide* (N0060600).	

Voice over IP trunks

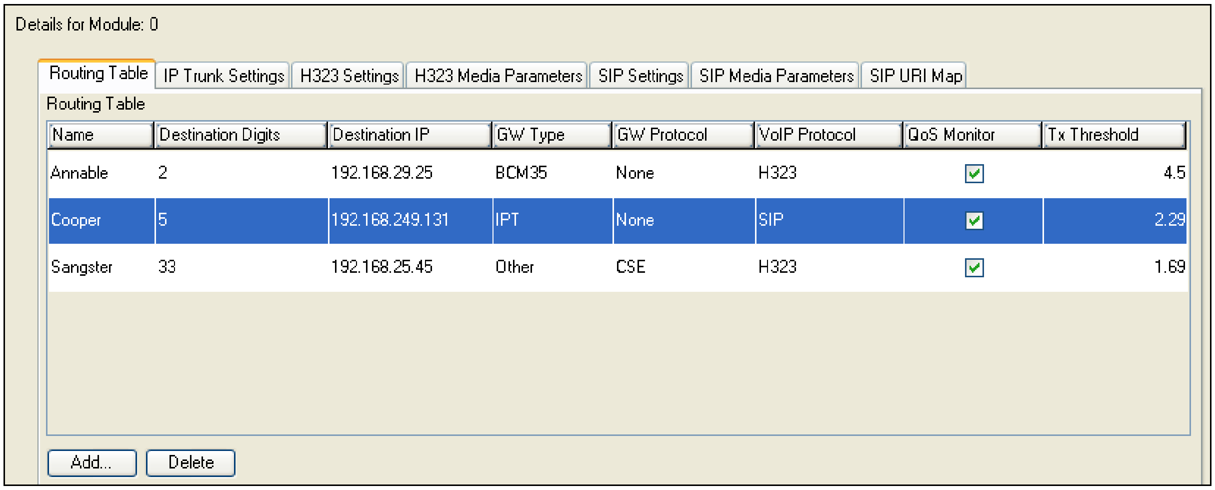
The tabbed panels described in the topics below appear when a VoIP trunk entry is selected on the Telephony Resources panel. Refer to the sections below for a description of each tabbed panel and their fields.

- "Routing table"
- "H323 Media Parameters" on page 140

Routing table

H.323 and SIP trunks are both automatically be assigned to line pool BlocA. The decision about whether a given call will be through SIP or H.323 is made from the information in the Routing Table. Calls may be routed directly from entries in the Routing Table, or may use the services of a redirect proxy or gatekeeper.

Figure 38 Routing Table



Name	Destination Digits	Destination IP	GW Type	GW Protocol	VoIP Protocol	QoS Monitor	Tx Threshold
Annable	2	192.168.29.25	BCM35	None	H323	☑	4.5
Cooper	5	192.168.249.131	IPT	None	SIP	☑	2.29
Sangster	33	192.168.25.45	Other	CSE	H323	☑	1.69

Table 32 Routing Table fields (Sheet 1 of 2)

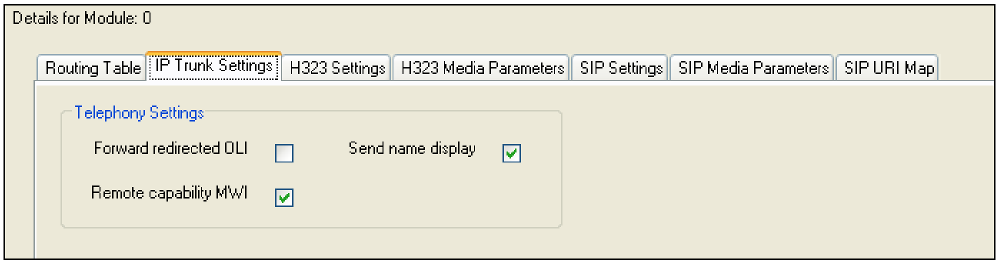
Attribute	Value	Description
Name	<alphanumeric>	Enter the name of the remote system
Destination Digits	<numeric> (could be the same as the destination code for the route to this system)	Set the leading digits which callers can dial to route calls through the remote gateway. Ensure that there are no other remote gateways currently using this combination of destination digits. If multiple leading digits map to the same remote gateway, separate them with a space. For example, 7 81 9555. These numbers are passed to the remote system as part of the dialed number.
Destination IP	<IP Address>	Enter the IP address of the remote system gateway.
GW Type	BCM BCM35 IPT Other	Choose the type of system that is accessed through the remote gateway: BCM: BCMs running 3.6 or newer software and CallPilot with compatible versions of H.323. BCM35: for BCMs running 3.5 software. IPT: Meridian 1 system running IP software.
GW Protocol	SL1 CSE None	Select the gateway protocol that the trunk expects to use. None: No special features. SL1: Use for BCM 2.5 systems only that require MCDN over VoIP trunks. CSE: MCDN protocol for gateways that provide VoIP service through BCM, Meridian 1 IPT, or CSE1000 gateways
VoIP Protocol	H323 SIP	Select the routing protocol for your network.
QoS Monitor	<check box>	If you intend to use a fallback PSTN line for this gateway, ensure that this check box is selected. Ensure that QoS Monitor is also enabled on the remote system. Otherwise, leave the check box empty.

Table 32 Routing Table fields (Sheet 2 of 2)

Attribute	Value	Description
Tx Threshold	<0-5>	Indicate the level of transmission at which the signal must be maintained. If the signal falls below this level the call falls back to PSTN. Default: 0
Actions		
Add	1. On the Remote Gateways panel, click **Add**. On the **Add** dialog: 2. **Name:** Enter a short descriptive title for the remote system. 3. **Destination IP**: Enter the public IP address of the remote system. 4. Click **OK**. 5. On the Remote Gateways panel, click in the fields to set any other parameters that you require.	
Delete	1. On the Remote Gateways panel, select the gateway you want to delete. 2. Click **Delete**. 3. Click **OK** on the confirmation dialog box.	

IP Trunk Settings

Figure 39 IP Trunk Settings fields



Details for Module: 0

Forward redirected OLI □

Send name display ☑

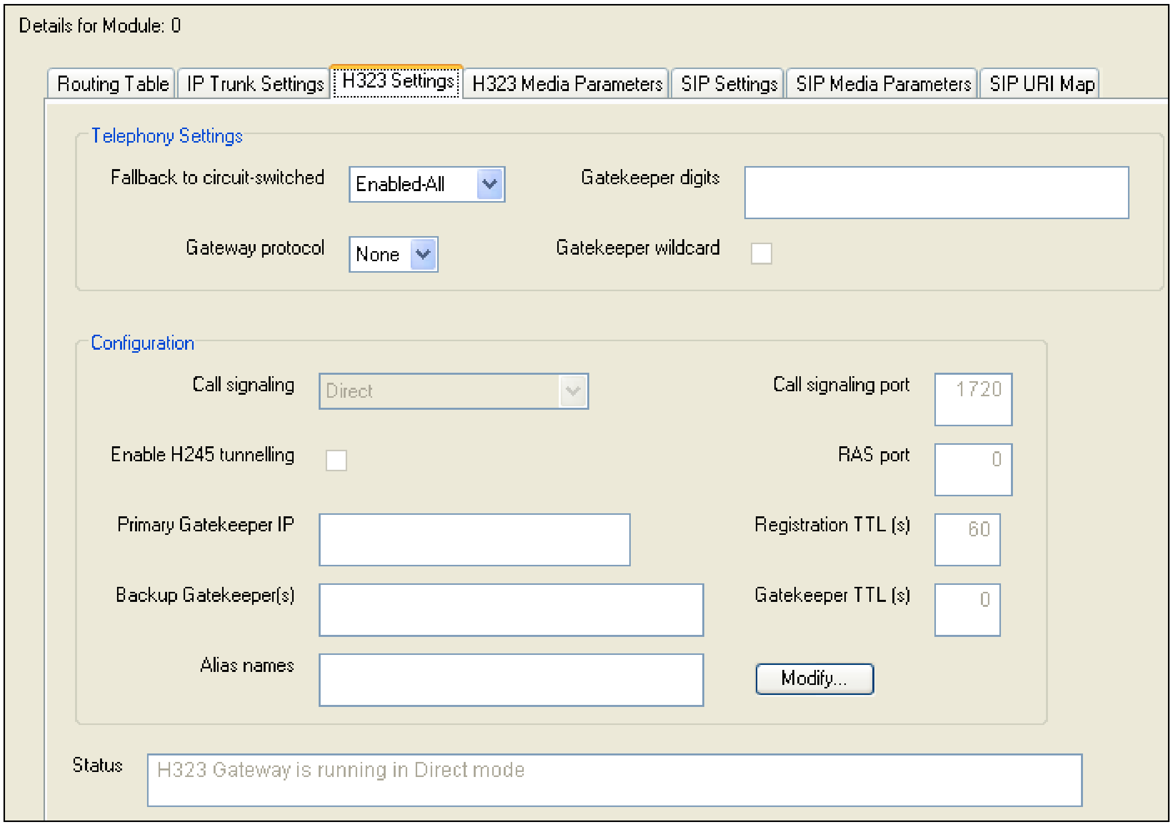
Remote capability MWI ☑

Table 33 IP Trunk Settings

Attribute	Value	Description
Telephony Settings		
Forward redirected OLI	<check box>	If the check box is selected, the OLI of an internal telephone is forwarded over the VoIP trunk when a call is transferred to an external number over the private VoIP network. If the check box is cleared, only the CLID of the transferred call is forwarded.
Remote capability MWI	<check box>	This setting must coordinate with the functionality of the remote system hosting the remote voice mail.
Send name display	<check box>	When selected, the telephone name is sent with outgoing calls to the network.

H323 Settings

Figure 40 H323 Settings


Details for Module: 0

Fallback to circuit-switched: Enabled-All

Gatekeeper digits:

Gateway protocol: None

Gatekeeper wildcard: □

Call signaling: Direct

Call signaling port: 1720

Enable H245 tunnelling: □

RAS port: 0

Primary Gatekeeper IP:

Registration TTL (s): 60

Backup Gatekeeper(s):

Gatekeeper TTL (s): 0

Alias names:

Status: H323 Gateway is running in Direct mode

Table 34 describes the fields on the H323 Settings tab.

Table 34 H323 Settings fields (Sheet 1 of 4)

Field	Value	Description
Telephony Settings		
Fallback to circuit-switched	Enabled-All Enabled-TDM Disabled	Your choice determines how the system will handle calls if the IP network cannot be used. • Enabled-All: All calls are rerouted over specified PSTN trunks lines. • Enabled-TDM: All TDM (digital telephones) voice calls will be rerouted over specified PSTN trunks lines. • Disabled: Calls will not be rerouted. Default: Enabled-All
	Note: Enabled-TDM-only enables fallback for calls originating on digital telephones. This is useful if your IP telephones are connected remotely, on the public side of the BCM network, because PSTN fallback is unlikely to result in better quality of service in that scenario.	

Table 34 H323 Settings fields (Sheet 2 of 4)

Field	Value	Description
Gateway protocol	None SL1 CSE	Both these protocols require a keycode. SL1: use this protocol only for BCM 2.5 systems CSE: Use this protocol for BCM 3.0 and newer systems. This protocol supports Meridian 1 IPT. Otherwise, use None.
Gatekeeper digits	<0-9>	If dialed digits match gatekeeper digits, the call is routed via H323 protocol. If the digits do not match, the call is routed via SIP protocol.
Gatekeeper wildcard	<check box>	If selected, all dialed digits match gatekeeper digits and VoIP calls will be routed through the gatekeeper.
Configuration		
*Call signaling	Direct Gatekeeper Resolved Gatekeeper Routed Gatekeeper Routed no RAS	Direct: call signaling information is passed directly between endpoints. The remote gateway table in the Element Manager defines a destination code (digits) for each remote system to direct the calls for that system to route. In each system, the Nortel IP Terminals and H.323 Terminals records map IP addresses to specific telephones. Gatekeeper Resolved: all call signaling occurs directly between H.323 endpoints. This means that the gatekeeper resolves the phone numbers into IP addresses, but the gatekeeper is not involved in call signaling. Gatekeeper Routed: uses a gatekeeper for call setup and control. In this method, call signaling is directed through the gatekeeper. Gatekeeper Routed no RAS: Use this setting for a NetCentrex gatekeeper. With this setting, the system routes all calls through the gatekeeper but does not use any of the gatekeeper Registration and Admission Services (RAS).
Enable H245 tunneling	<check box>	If Enabled, the VoIP Gateway tunnels H.245 messages within H.225. The VoIP Gateway service must be restarted for a change to take effect. Default: Disabled.
Primary Gatekeeper IP	<IP address>	If Gatekeeper Routed, Gatekeeper Resolved or Gatekeeper Routed no RAS are selected under Call Signaling, type the IP address of the machine that is running the gatekeeper.
Backup Gatekeeper(s)	<IP address>, <IP address>	NetCentrex gatekeeper does not support RAS, therefore, any backup gatekeepers must be entered in this field. **Note:** Gatekeepers that use RAS can provide a list of backup gatekeepers for the end point to use in the event of the primary gatekeeper failure.

Table 34 H323 Settings fields (Sheet 3 of 4)

Field	Value	Description
Alias Names	Alias names are comma delimited, and may be one of the following types: **E.164** — numeric identifier containing a digit in the range 0-9. Identified by the keyword `TEL:` Example: the BCM is assigned an E.164 and an H323 Identifier: `Alias Names: TEL:76, NAME:bcm10.nortel.com` • NPI-TON — also referred to as a PartyNumber alias. Similar to E164 except that the keyword indicates the NPI (numbering plan identification), as well as the TON (type of number). Identified by one of the following keywords: `PUB` (Public Unknown Number); `PRI` (Private Unknown Number); `UDP` (Private Level 1 Regional Number (UDP)); `CDP` (Private Local Number (CDP)).	
	• H.323Identifier — alphanumeric strings representing names, e-mail addresses, etc. Identified by the keyword `NAME:` Example: The BCM is assigned a public dialed number prefix of 76, a private CDP number of 45, and an H323 Identifier alias: `Alias Names: PUB:76, CDP:45, NAME:bcm10.nortel.com`	
	• H.225 (Q.931) CallingPartyNumber (NetCentrex gatekeeper) — The NetCentrex gatekeeper uses the H.225(Q.931) CallingPartyNumber to resolve the call originator for billing purposes. This number must then contain a unique prefix, or location code that is unique across all endpoints that are using the NetCentrex gatekeeper. Identified by the keyword **`src:`**. Example for private networks: CDP alias = src:<DN>; UDP alias = src:<LOC><DN>. Example for public network: src:<public OLI>	
	Note: E164 or NPI-TON alias types are commonly used since they fit into dialing plans. A BCM alias list should not mix these types. Also, the type of alias used should be consistent with the dialing plan configuration. Use the same alias naming method on all BCMs within a network.	
Configuration note:	Refer to “Using CS 1000 as a gatekeeper” on page 417 for specific information about configuring the gatekeeper for H.323 trunks. Network note: If your private network contains a Meridian 1-IPT, you cannot use Radvision for a gatekeeper.	
If Gatekeeper Routed, Gatekeeper Resolved, or Gatekeeper Routed no RAS are selected under Call Signaling, enter one or more alias names for the gateway.		
Call signaling port	0-65535	Default: 1720 This field allows you to set non-standard call signaling port for VoIP applications that require special ports. 0 = The first available port is used. Ensure that you do not select a port that has been assigned elsewhere in the BCM. To ensure the port is not in use, run netstat-a from the command line.
RAS port	0-65535	Default: 0 This field allows you to set a non-standard Registration and Admission (RAS) port for VoIP applications that require special ports. 0 = The first available port is used. Ensure that you do not select a port that has been assigned elsewhere in the BCM. To ensure the port is not in use, run netstat-a from the command line.
Registration TTL (s)	Default: 60 seconds	This TimeToLive parameter specifies the intervals when the VoIP gateway sends KeepAlive signals to the gatekeeper. The gatekeeper can override this timer and send its own TimeToLive period.

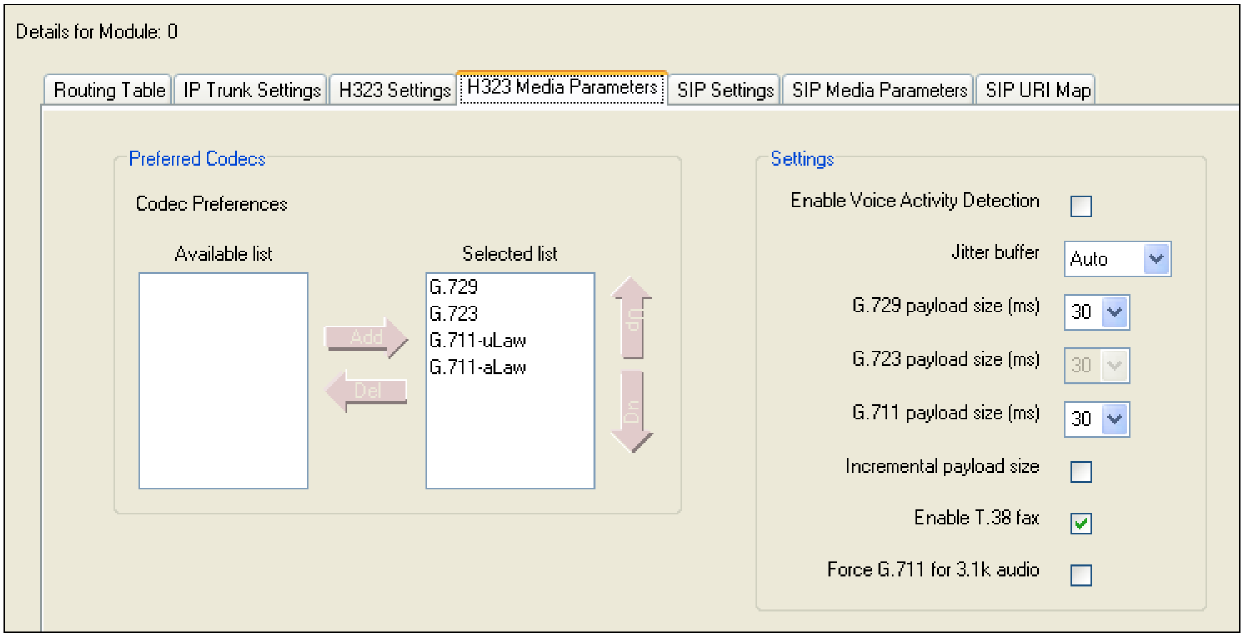
Table 34 H323 Settings fields (Sheet 4 of 4)

Field	Value	Description
Gatekeeper TTL (s)		The actual time used by the gatekeeper for the registration process.
Status	<read-only>	Indicates if the device is online.
Modify	<button>	Click to modify the parameters. **Note**: All active H.323 calls are dropped if these settings are changed.

H323 Media Parameters

The H323 Media Parameters tab controls codec settings for H323 trunks. This panel also includes the settings to enable T.38 Fax signals over the trunks.

Figure 41 H323 Media Parameters panel



Details for Module: 0

Selected list: G.729, G.723, G.711-uLaw, G.711-aLaw

Enable Voice Activity Detection □

Jitter buffer: Auto

G.729 payload size (ms): 30

G.723 payload size (ms): 30

G.711 payload size (ms): 30

Incremental payload size □

Enable T.38 fax ☑

Force G.711 for 3.1k audio □

Table 35 describes the fields on this panel.

Table 35 H323 Media parameters record (Sheet 1 of 3)

Field	Value	Description
Preferred Codecs		
Preferred Codecs	None G.711-uLaw G.711-aLaw G.729 G.723	Select the Codecs in the order in which you want the system to attempt to use them. **Performance note:** Codecs on all networked BCMs must be consistent to ensure that interacting features such as Transfer and Conference work correctly. Systems running BCM 3.5 or newer software allow codec negotiation and renegotiation to accommodate inconsistencies in Codec settings over VoIP trunks.

Table 35 H323 Media parameters record (Sheet 2 of 3)

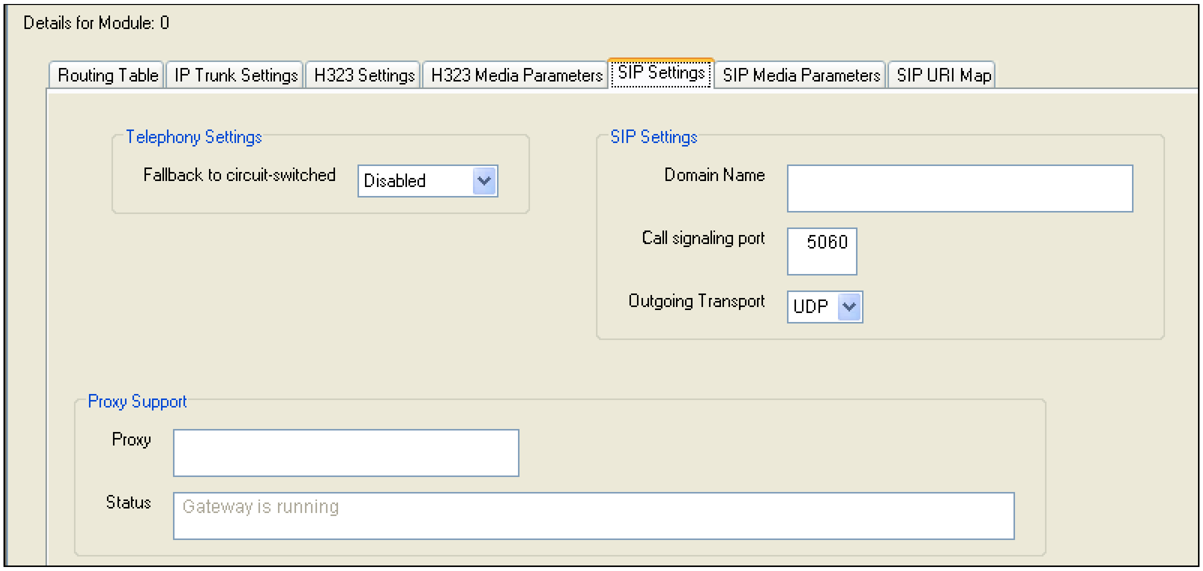
Field	Value	Description
Actions		
→	1. On the Available list, click the codec you want to add to the Selected list. 2. Click the button to move the codec to the Selected list.	
←	1. Select a codec that you want to remove from the Selected list. 2. Click this button to move the codec back to the Available list.	
↑ ↓	1. Select a codec on the Selected list. 2. Click the appropriate arrow to move the codec up or down in the Selected list.	
Settings		
Enable Voice Activity Detection	<check box>	Voice activity detection, also known as silence suppression identifies periods of silence in a conversation, and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. For more information refer to "Silence suppression" on page 695. If voice activity detection is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support voice activity detection. G.711 does not support voice activity detection. **Performance note:** Voice activity detection on all networked BCMs and IPT systems (VAD setting on IPT systems) must be consistent to ensure that interacting features such as Transfer and Conference work correctly. As well, the Payload size on the IPT must be set to 30ms. Default: Disabled
Jitter buffer	Auto None Small Medium Large	Select the size of jitter buffer you want to allow for your system. Default: Auto
G.729 payload size (ms)	10, 20, 30, 40, 50, 60 Default: 30	Set the maximum required payload size, per codec, for the VoIP calls sent over H.323 trunks. **Note:** Payload size can also be set for Nortel IP telephones. Refer to the BCM 4.0 Telephony Device Installation Guide (N0027269).
G.723 payload size (ms)	30	
G.711 payload size (ms)	10, 20, 30, 40, 50, 60 Default: 30	
Incremental payload size	<check box>	When enabled, the system advertises a variable payload size (40, 30, 20, 10 ms)
Enable T.38 fax	<check box>	Enabled: The system supports T.38 fax over IP. Disabled: The system does not support T.38 fax over IP

Table 35 H323 Media parameters record (Sheet 3 of 3)

Field	Value	Description
	⛔	**Caution: Operations note:** Fax tones that broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. Here are some suggestions to minimize the possibility of your VoIP calls being dropped because of fax tone interference: Locate fax machine away from other telephones. Turn the speaker volume on the fax machine to the lowest level, or off, if that option is available.
Force G.711 for 3.1k Audio	<check box>	When enabled, the system forces the VoIP trunk to use the G.711 codec for 3.1k audio signals such as modem or TTY machines. **Note**: This setting can also be used for fax machines if T.38 fax is not enabled on the trunk.

SIP Settings

Figure 42 SIP Settings tab



Details for Module: 0

Fallback to circuit-switched: Disabled

Domain Name:

Call signaling port: 5060

Outgoing Transport: UDP

Proxy:

Status: Gateway is running

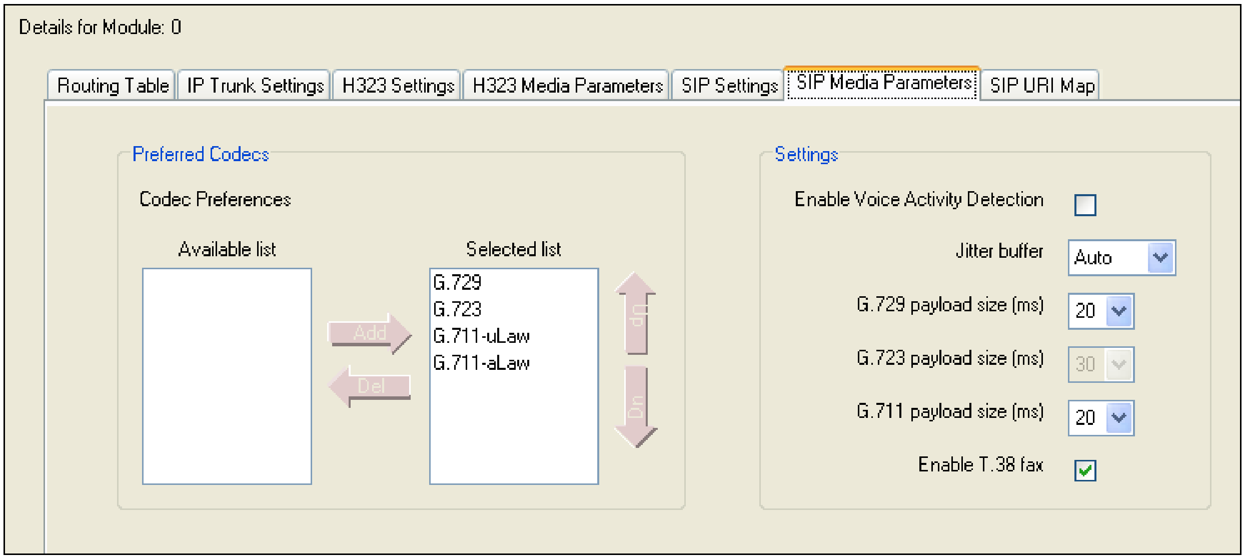
Table 36 SIP Settings fields

Field	Value	Description
Telephony Settings		
Fallback to circuit-switched	Enabled-All Enabled-TDM Disabled	Your choice determines how the system will handle calls if the IP network cannot be used. • Enabled-All: All calls will be rerouted over specified PSTN trunks lines. • Enabled-TDM: All TDM (digital telephones) voice calls will be rerouted over specified PSTN trunks lines. • Disabled: Calls will not be rerouted. Default: Enabled-All
SIP Settings		
Domain Name		Domain of the SIP network.
Call signaling port	<numeric>	The listening port for the BCM. **Note**: FEPS (Functional Endpoint Proxy Server) must be restarted if this value is changed. Default: 5060
Outgoing Transport	UDP TCP	Select the outgoing transport protocol for the gateway. **Note**: UDP is the only transport supported by the SIP enabled data services. Default: UDP
Proxy Support		
Proxy	<IP address>	Specify the IP address of the SIP proxy server.
Status	<read-only>	Indicates the status of the gateway.

SIP Media Parameters

SIP trunks are administered separately from H.323 trunks. It is common for H.323 and SIP trunks to both exist on the same system; however, each has different network segments.

Figure 43 SIP Media Parameters tab



Details for Module: 0

Preferred Codecs

Codec Preferences

Available list

Selected list: G.729, G.723, G.711-uLaw, G.711-aLaw

Settings

Enable Voice Activity Detection □

Jitter buffer: Auto

G.729 payload size (ms): 20

G.723 payload size (ms): 30

G.711 payload size (ms): 20

Enable T.38 fax ☑

Table 37 SIP Media parameters tab (Sheet 1 of 2)

Field	Value	Description
Preferred Codecs		
Preferred Codecs	None G.711-uLaw G.711-aLaw G.729 G.723	Select the Codecs in the order in which you want the system to attempt to use them. **Performance note:** Codecs on all networked BCMs should be consistent to ensure that interacting features such as Transfer and Conference work correctly. **Note**: The G.723 protocol can be used between IP endpoints.
Actions		
→	1. On the Available list, click the codec you want to add to the Selected list. 2. Click the button to move the codec to the Selected list.	
←	1. Select a codec that you want to remove from the Selected list. 2. Click this button to move the codec back to the Available list.	
↑ ↓	1. Select a codec on the Selected list. 2. Click the appropriate arrow to move the codec up or down in the Selected list.	

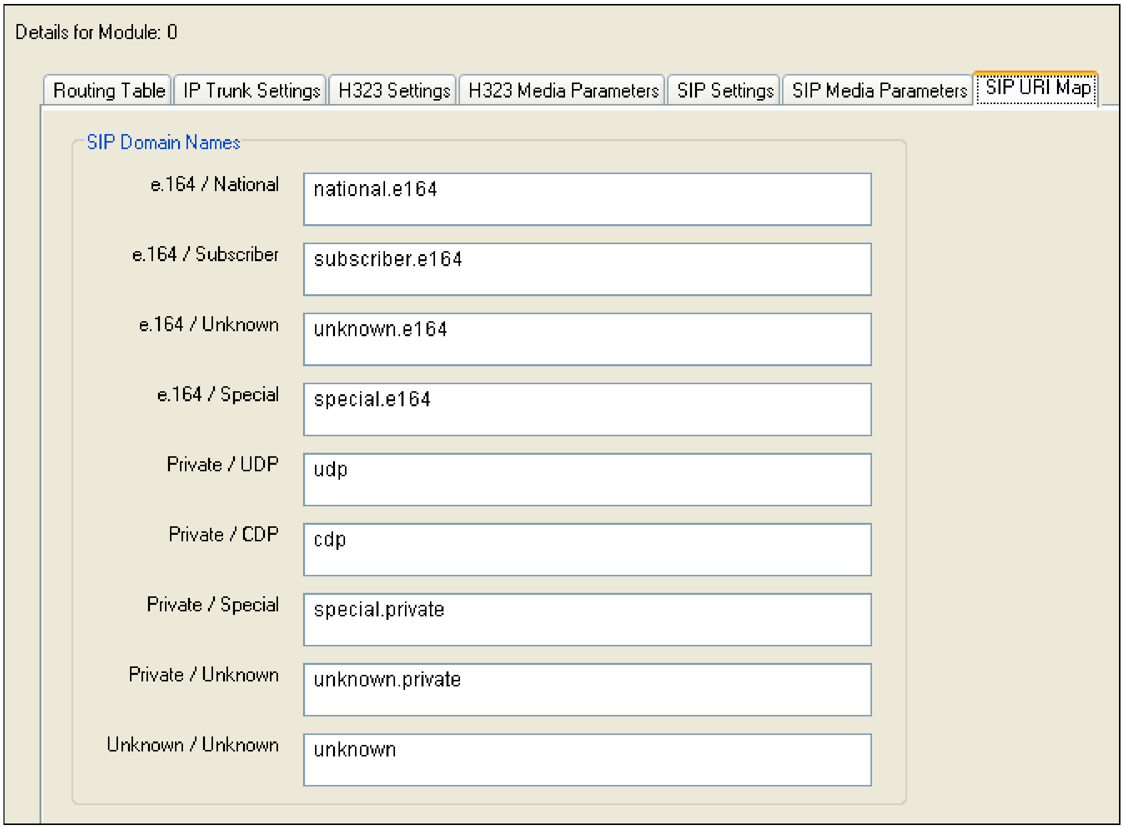
Table 37 SIP Media parameters tab (Sheet 2 of 2)

Field	**Value**	**Description**
Settings		
Enable Voice Activity Detection	<check box>	The voice activity detection (silence suppression) identifies periods of silence in a conversation, and stops sending IP speech packets during those periods. In a typical telephone conversation, most of the conversation is half-duplex, meaning that one person is speaking while the other is listening. For more information refer to "Silence suppression" on page 695. If voice activity detection is enabled, no voice packets are sent from the listener end. This greatly reduces bandwidth requirements. G.723.1 and G.729 support silence suppression. G.711 does not support silence suppression. **Performance note:** voice activity detection on all networked BCMs and IPT systems (VAD setting on IPT systems) must be consistent to ensure that interacting features such as Transfer and Conference work correctly. Default: Disabled
Jitter Buffer	Auto None Small Medium Large	Select the size of jitter buffer you want to allow for your system.
G.729 Payload Size (ms)	10, 20, 30, 40, 50, 60 Default: 30	Set the desired payload size, per codec, for the VoIP calls sent over SIP trunks. **Note:** Payload size can also be set for Nortel IP telephones. Refer to the *BCM 4.0 Telephony Device Configuration Guide* (N0027269).
G.723 Payload Size (ms)	30	
G.711 Payload Size (ms)	10, 20, 30, 40, 50, 60 Default: 30	
Enable T.38	<check box>	Enabled: The system supports T.38 fax over IP. Disabled: The system does not support T.38 fax over IP
	⛔	**Caution: Operations note:** Fax tones that broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. Here are some suggestions to minimize the possibility of your VoIP calls being dropped because of fax tone interference: Locate fax machine away from other telephones. Turn the speaker volume on the fax machine to the lowest level, or off, if that option is available.

SIP URI Map

Use the SIP URI Map to configure the sub-domain name associated with each SIP URI (Session Initiated Protocol Uniform Resource Identifier). These strings must be coordinated with the other nodes in the network so that the type of dialing.

Figure 44 SIP URI Map tab



Details for Module: 0

SIP Domain Names

e.164 / National: national.e164

e.164 / Subscriber: subscriber.e164

e.164 / Unknown: unknown.e164

e.164 / Special: special.e164

Private / UDP: udp

Private / CDP: cdp

Private / Special: special.private

Private / Unknown: unknown.private

Unknown / Unknown: unknown

Table 38 SIP URI Map Fields

Field	Value	Description
SIP Domain Names		
e.164 / National	national.e164	String to use in phone context to identify numbering plan type
e.164 / Subscriber	subscriber.e164	String to use in phone context to identify numbering plan type
e.164 / Special	special.e164	String to use in phone context to identify numbering plan type
e.164 / Unknown	unknown.e164	String to use in phone context to identify numbering plan type
Private / UDP	UDP	String to use in phone context to identify numbering plan type
Private / CDP	CDP	String to use in phone context to identify numbering plan type
Private / Special	special.private	String to use in phone context to identify numbering plan type
Private / Unknown	unknown.private	String to use in phone context to identify numbering plan type
Unknown / Unknown	unknown	String to use in phone context to identify numbering plan type

Chapter 9
Configuring lines

All the Lines panels show the same type of tabbed panels. The information on the tabbed panels may vary, however, depending on the type of line.

The following paths indicate where to access the lines information in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Tallest interface: ****CONFIG > Lines**

The top panel provides a table of lines and the current or default settings.

The bottom frame contains up to five tabs. The contents of the tabs may vary, depending on the line selected in the top table.

- The Parameters tabbed panel provides general information on the selected DN.
- The Properties and Preferences tabbed panels provide the settings for individual line characteristics.
- The Restrictions tabbed panel allows you to define which restrictions will be active for individual lines. Note that lines that are assigned to the same line pool will automatically assign the same restrictions.
- The Assigned DNs tabbed panel provides a quick way to assign lines to telephones. You must use the DN records panels to assign line pools to telephones.

Click one of the following links to access the information you want to view:

Panel tabs	Tasks
“Trunk/Line data - main panel” on page 148	“Configuring lines: T1-Loop start” on page 187
“Properties” on page 151	“Configuring lines: T1-Digital Ground Start” on page 193
“Restrictions (Line and Remote)” on page 156	“Configuring lines: T1-E&M” on page 181
“Assigned DNs” on page 157 See also: Line Access - Line Assignment tab in the *BCM 4.0 Device Configuration Guide* (N0060600)	“Configuring lines: T1-DID” on page 199
	“Configuring lines: PRI” on page 171
	“Configuring lines: DPNSS lines” on page 211
	“Configuring lines: Target lines” on page 177
	“Configuring BRI lines” on page 219
	“Configuring VoIP lines” on page 414
	“Call Security: Configuring Direct Inward System Access (DISA)” on page 451

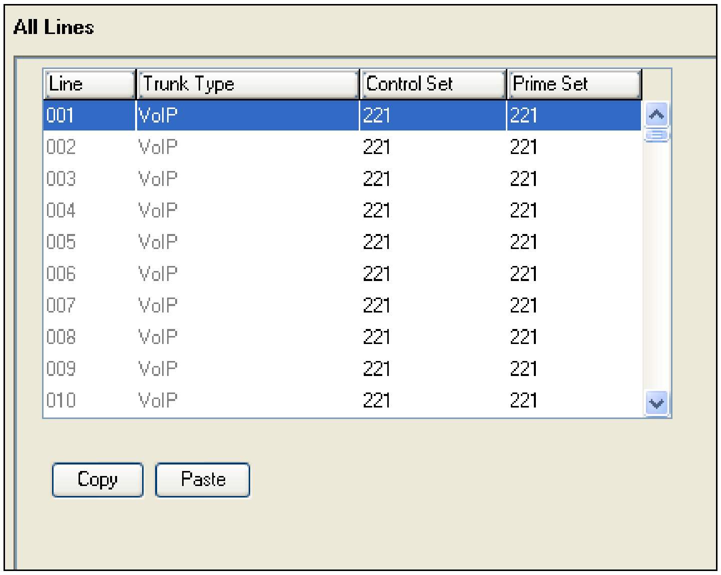
Click the navigation tree heading to access general information about user management.

Note: Not all the fields shown below necessarily appear for any one type of line. Some fields relate to specific lines.

Trunk/Line data - main panel

The top-level Table View panel shows line records for all lines active on the system, and the common assigned parameters. Figure 45 shows the Trunk/Line Data lines panel.

Figure 45 Trunk/Line Data lines panel



All Lines

Line	Trunk Type	Control Set	Prime Set
001	VoIP	221	221
002	VoIP	221	221
003	VoIP	221	221
004	VoIP	221	221
005	VoIP	221	221
006	VoIP	221	221
007	VoIP	221	221
008	VoIP	221	221
009	VoIP	221	221
010	VoIP	221	221

- Also refer to “Common procedures: copying and renumbering DNs” in the *BCM 4.0 Device Configuration Guide* (N0060600).

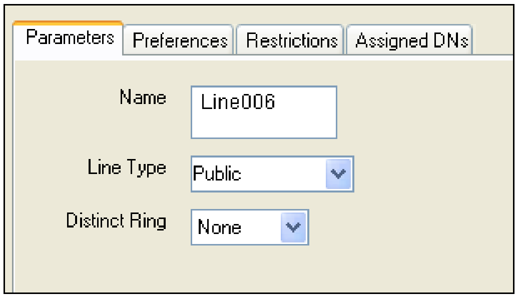
Table 39 describes the fields found on the Trunk/Line Data main panel.

Table 39 Trunk/Line Data main panel

Attribute	Value	Description
Line	This list contains all the possible line numbers for the system, including target lines.	Configure only those lines that are active on the system. (Click the Active check box and ensure that the Inactive check box is empty).
Trunk Type	Loop, PRI, VoIP, Target	There are three main categories of lines: PSTN-based lines: (analog, T1, PRI, BRI) Voice over IP (VoIP) trunks, which connect through the LAN or WAN. Target lines, which are internal channels that provide direct dial capability.
Control Set	DN <control telephone DN> Default: 221 (default Start DN)	Enter a telephone DN for a telephone that you want to use to turn service off or on for other telephones using this line. The control telephone must have the line assigned, or must be assigned to the line pool the line is in. Refer to “Line Access - Line Pool Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).
➡	**Tips:** External lines and telephones must be programmed to use one of the Scheduled Services: Ringing, Restriction, and Routing Services. For maximum flexibility, Nortel recommends that you create two different control telephones, one for the lines and one for the telephones. You can turn on a service manually or automatically for all external lines from an assigned control telephone. However, you cannot combine schedules. A service can only be active as normal service or one of the six schedules at any one time. Several schedules can be active at one time, but they must use different services.	
Prime Set	DN: None	Assign a telephone to provide backup answering for calls on the line. For an Auto Answer line, calls are redirected if the received number is invalid or the target line is busy, and if the **If busy** parameter is set **To prime**. Each line can be assigned only one prime telephone. Doorphone note: Ensure that this DN does not belong to a doorphone.

Parameters

Figure 46 Parameters details tab



Name: Line006

Line Type: Public

Distinct Ring: None

Table 3 Parameters details tab (Sheet 1 of 2)

Attribute	**Value**	**Description**
Name	<maximum of seven alphanumeric characters>	Identify the line in a way that is meaningful to your system, such as by the type of line and line pool or the DN it is attached to in the case of target lines.
Line Type	Public Private to: Pool A to O, BlocA to BlocF	Define how the line is used in relation to other lines in the system. • Public line: can be accessed by more than one telephone. • Private line: can be assigned only to one telephone and the prime telephone for that line. Enter the internal number of the telephone. • Pool A - O (digital lines and VoIP lines) • BlocA to BlocF (PRI and ETSI QSIG lines): assigns the line to one of the line pools. If a line is assigned to a line pool, but is not assigned to any telephone, that line is available only for outgoing calls. Bloc line pools must be used in conjunction with routes and destination codes. Target lines cannot be put into line pools.
Pub. Received # (Target lines only)	<digits associated with a specific target line>	Specify the digits the system will use to identify a call from the public network to this target line. • A received number cannot be the same as, or be the start digits, of a line pool access code, a destination code, the DISA DN or the Auto DN. • If you are configuring auto-answer BRI trunks to map to target lines, the received number should be the same as the Network DN supplied by your service provider. The call will be directed to the prime telephone for the incoming line if the Network DN is not used.

Table 3 Parameters details tab (Sheet 2 of 2)

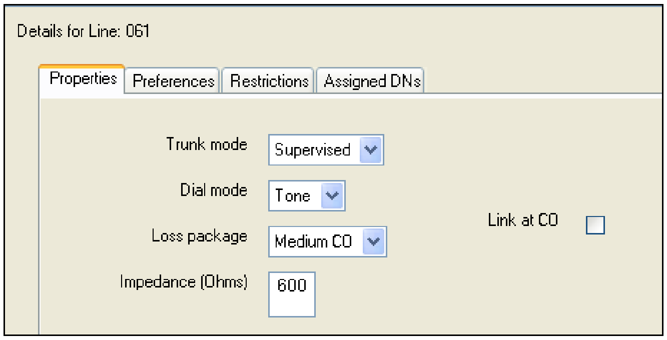
Attribute	Value	Description
Priv. Received # (Target lines only)	<digits associated with a specific target line>	Specify the digits the system will use to identify a call from the private network to this target line. • A received number cannot be the same as, or be the start digits, of a line pool access code, a destination code, the DISA DN or the Auto DN. • If you are configuring auto-answer BRI trunks to map to target lines, the received number should be the same as the Network DN supplied by your service provider. The call will be directed to the prime telephone for the incoming line if the Network DN is not used.
Distinct ring	None Pattern 2 Pattern 3 Pattern 4	Choose the distinctive ring pattern that you want to assign to the line. This allows you to provide selective service to calls with differing answer priorities. When more than one line with the distinct ring settings rings at a telephone, the line with the highest priority will ring first. • Pattern 4 has the highest ring priority • Pattern 3 has second highest ring priority • Pattern 2 has third highest ring priority • None has the lowest ring priority. By default, all telephones and lines are set to None.

Properties

The Properties tab shows basic line properties. Not all fields apply to all types of lines.

The Properties tab is shown in Figure 47.

Figure 47 Properties details tab



Details for Line: 061

Trunk mode: Supervised

Dial mode: Tone

Loss package: Medium CO

Impedance (Ohms): 600

Link at CO □

Table 40 defines the fields on this panel and indicates the lines.

Table 40 Properties line settings (Sheet 1 of 2)

<table>
<tr><th>Attribute</th><th colspan="2">Value</th><th colspan="6">Description</th></tr>
<tr><td colspan="9"><b>Legend: Loop = analog/digital loop; GS = ground start; DID = DID; E&M = E&M; BRI = BRI; DPNSS = DPNSS; VoIP = VoIP; TL = Target. Note: PRI fields are all included under the main table.</b></td></tr>
<tr><td rowspan="2">Trunk mode</td><td><b>Loop</b></td><td></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Unspr
Supervised
*Earth calling
*Loop guarded
*Loop unguarded</td><td colspan="6">Define whether disconnect supervision, also referred to as loop supervision, releases an external line when an open switch interval (OSI) is detected during a call on that line. You must set this to Supervised if a loop trunk has its Answer mode set to Auto or if you enable Answer with DISA. Disconnect supervision is also required to conference two external callers. The line must be equipped with disconnect supervision from the central office for the Supervised option to work.
* These listing only appear for UK analog lines.</td></tr>
<tr><td rowspan="2">Dial mode</td><td><b>Loop</b></td><td><b>GS</b></td><td><b>DID</b></td><td><b>E&M</b></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Pulse
Tone</td><td colspan="6">Specify whether the system uses dual tone multifrequency (DTMF) or pulse signaling on the trunk.
Tone does not appear if Signaling is set to Immediate (T1 DID &T1 E&M trunk types only).</td></tr>
<tr><td rowspan="2">Loss package</td><td colspan="2"><b>Loop (analog only)</b></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">Short CO
Medium CO
Long CO
Short PBX
Long PBX</td><td colspan="6">Select the appropriate loss/gain and impedance settings for each line, see Table 46.</td></tr>
<tr><td rowspan="2">Impedance (Ohms)</td><td colspan="2"><b>Loop (analog only)</b></td><td></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">600 ohm
900 ohm</td><td colspan="6">The GATM can be set to a specific impedance level.</td></tr>
<tr><td rowspan="2">Signaling</td><td></td><td><b>DID</b></td><td><b>E&M</b></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">WinkStart
Immediate
DelayDial</td><td colspan="6">Select the signal type for the line. The immediate setting does not appear for T1 E&M or T1 DID trunks connected to a DTM if the Dial mode is set to tone.
Make sure that this matches the signal type programmed for the trunk at the other switch.</td></tr>
</table>

Table 40 Properties line settings (Sheet 2 of 2)

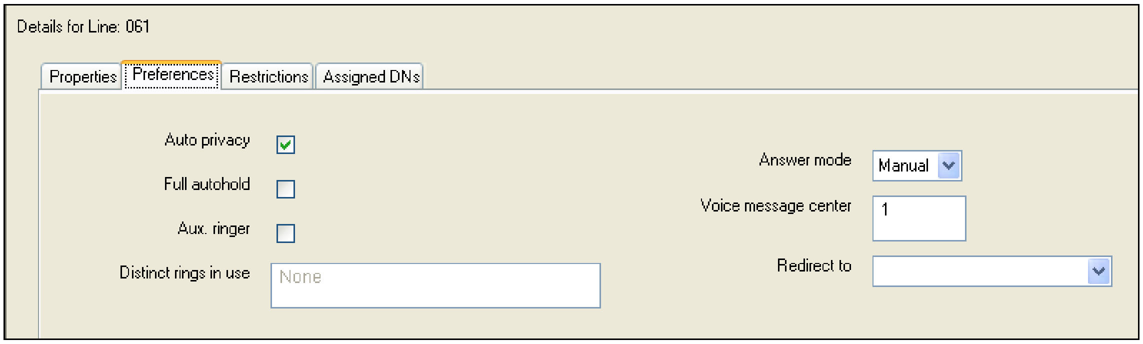
Attribute	Value	Description					
Legend: Loop = analog/digital loop; GS = ground start; DID = DID; E&M = E&M; BRI = BRI; DPNSS = DPNSS; VoIP = VoIP; TL = Target. Note: PRI fields are all included under the main table.							
Link at CO	**Loop (analog only)**						
	<check box>	Some exchanges respond to a Link signal, also called hook flash (**FEATURE 71**), by providing an alternative line for making outgoing calls. Enabling Link at CO causes the system to apply the restrictions on outgoing calls to the digits dialed after the Link signal. As well, the call on the alternative line is subject to all restrictions. Disabling Link at CO prevents a Link signal from resetting the BCM restrictions in cases where the host exchange does not provide an alternative line.					
Link time	**Loop (analog only)**						
	<time>	Link at CO is enabled. The duration of the on-hook signal sent when the user activates the Link feature.					
Dial tone (detect delay)	**Loop (analog only)**						
	Detect Delay	This field tells the system to either detect a dial tone before sending the dialstring, or to wait a period of time and then send the dial string.					

Preferences (lines)

The Preferences tab shows information that may vary from trunk to trunk. Most of this information needs to coordinate with the line service provider equipment.

Figure 48 shows the Preferences tab.

Figure 48 Preferences details panel



Details for Line: 061

Auto privacy ☑

Full autohold □

Aux. ringer □

Distinct rings in use: None

Answer mode: Manual

Voice message center: 1

Redirect to:

Table 41 defines the fields on this panel and indicates the lines.

Table 41 Preferences details fields for lines (Sheet 1 of 3)

Attribute	**Value**		**Description**					
Legend: Loop = analog/digital loop; GS = ground start; DID = DID; E&M = E&M; BRI = BRI; DPNSS = DPNSS; VoIP = VoIP; TL = Target and DASS2. Note: PRI fields are all included under the main panel.								
Auto privacy	**Loop**	**GS**	**DID**	**E&M**	**BRI**		**VoIP**	
	<check box>		Define whether one BCM user can select a line in use at another telephone to join an existing call. Refer to “Turn Privacy on or off” in the *BCM 4.0 Device Configuration Guide* (N0060600) (**FEATURE 83**).					
Full autohold	**Loop**				**BRI**	**DPNSS**	**VoIP**	
	<check box>		Enables or disables Full autohold. When enabled, if a caller selects an idle line but does not dial any digits, that line is automatically placed on hold if you then select another line. Full autohold is always in place for T1 E&M trunks because it has no meaning for incoming-only T1 DID trunks. The default setting should be changed only if Full autohold is required for a specific application.					
Aux. ringer	**Loop**	**GS**	**DID**	**E&M**	**BRI**	**DPNSS**	**VoIP**	**TL**
	<check box>		Turn the auxiliary ringer on or off for all telephones using this line. When programmed on a line, the auxiliary ringer will ring every time a call is received.					
	Note: When programmed only on a telephone, no ring occurs for a transferred call. An auxiliary ringer can also be programmed in Services to ring for a line placed into a scheduled Ringing service. Refer to “Configuring scheduled service” in the *BCM 4.0 Device Configuration Guide* (N0060600).							
ANI Number		**DID**	**E&M**					
	<check box>		Define whether the telephone number of the caller will be shown for this line. For T1 E&M and T1 DID trunks connected to a DTM, this setting only appears if Signaling is set to WinkStart. The central office must deliver ANI/DNIS in DTMF mode. No additional equipment is required.					

Table 41 Preferences details fields for lines (Sheet 2 of 3)

<table>
<tr><th>Attribute</th><th colspan="2">Value</th><th colspan="6">Description</th></tr>
<tr><td colspan="9"><b>Legend: Loop = analog/digital loop; GS = ground start; DID = DID; E&M = E&M; BRI = BRI; DPNSS = DPNSS; VoIP = VoIP; TL = Target and DASS2. Note: PRI fields are all included under the main panel.</b></td></tr>
<tr><td rowspan="2">DNIS Number</td><td></td><td></td><td><b>E&M</b></td><td></td><td></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">&lt;check box&gt;</td><td colspan="6">Defines whether the digits dialed by an external caller on this line will be shown.For T1 E&M trunks connected to a DTM, this setting only appears if Signaling is set to WinkStart and Answer mode is set to Manual.</td></tr>
<tr><td>Distinct Rings in use</td><td colspan="2">&lt;read-only&gt;</td><td colspan="6">Indicates if a special ring has been assigned. See Distinct Ring on the main table.</td></tr>
<tr><td rowspan="3">Answer mode</td><td><b>Loop</b></td><td><b>GS</b></td><td></td><td><b>E&M</b></td><td><b>BRI</b></td><td><b>DPNSS</b></td><td></td><td></td></tr>
<tr><td colspan="2">Manual
Auto</td><td colspan="6">Define whether a trunk is manual or automatic answer.
Auto answer mode allows the trunk to be a shared resource by the system telephones. This shared resource is created through routing to target lines or using DISA.
For auto answer trunks being used to allow remote call-in from system users, the trunk can be configured to answer with a straight dial tone, if DISA has not been enabled. It can also be configured to answer with a stuttered dial tone if DISA is enabled and the caller is expected to enter a CoS password. The CoS password defines which system features the caller is permitted to access.
Manual answer trunks are assigned to one or more telephones. The assigned telephones exclusively own the line.</td></tr>
<tr><td colspan="8"><b>Note:</b> You require Disconnect supervision on the line if loop start trunks are to operate in auto-answer mode.</td></tr>
<tr><td rowspan="2">Answer with DISA</td><td><b>Loop</b></td><td><b>GS</b></td><td></td><td><b>E&M</b></td><td><b>BRI</b></td><td></td><td></td><td></td></tr>
<tr><td colspan="2">&lt;check box&gt;</td><td colspan="6">Define whether the system prompts a caller for a six-digit class of service (CoS) password. This setting appears for T1 loop start, T1 E&M lines that have auto-answer mode, and analog trunks. Set this option to No for T1 E&M lines on a private network that have auto-answer mode.
To program DISA on a PRI trunk you need to specify a DISA DN, see “Call Security: Configuring Direct Inward System Access (DISA)” on page 451 and “Dialing plan: Private network settings” on page 317.</td></tr>
<tr><td rowspan="3">If busy</td><td></td><td></td><td></td><td></td><td></td><td></td><td></td><td><b>TL</b></td></tr>
<tr><td colspan="2">To Prime
Busy Tone</td><td colspan="6">Define whether a caller receives a busy tone or the call forwards to the prime telephone when the target line is busy. Busy tone only works for PRI trunks.</td></tr>
<tr><td colspan="8"><b>Tips:</b> The duration of an open switch interval (OSI) before BCM disconnects a call is programmed by the Disconnect timer setting. Refer to “Trunk Module Parameters” on page 122.</td></tr>
<tr><td rowspan="2">Voice Message Center</td><td><b>Loop</b></td><td><b>GS</b></td><td><b>DID</b></td><td><b>E&M</b></td><td><b>BRI</b></td><td><b>DPNSS</b></td><td><b>VoIP</b></td><td><b>TL</b></td></tr>
<tr><td colspan="2">Center 1 - Center 5</td><td colspan="6">If this line connects t o a remote voice mail, either through the private network or at the Central Office, indicate which Center number has been configured with the contact number.The system calls that number to check voice mail messages when a message indicator is presented to a telephone.</td></tr>
</table>

Table 41 Preferences details fields for lines (Sheet 3 of 3)

<table>
<tr><th>Attribute</th><th colspan="2">Value</th><th colspan="6">Description</th></tr>
<tr><td colspan="9"><b>Legend: Loop = analog/digital loop; GS = ground start; DID = DID; E&M = E&M; BRI = BRI; DPNSS = DPNSS; VoIP = VoIP; TL = Target and DASS2. Note: PRI fields are all included under the main panel.</b></td></tr>
<tr><td rowspan="2">Redirect to</td><td><b>Loop</b></td><td><b>GS</b></td><td><b>DID</b></td><td><b>E&M</b></td><td></td><td></td><td></td><td><b>TL</b></td></tr>
<tr><td colspan="2">&lt;dial string&gt;</td><td colspan="6">Enter a dial string (including destination code) to redirect the line to an external telephone, such as a call attendant on another system.
If you want to stop redirection, you need to delete the dial string and allow the record to update.
<b>Warning</b>: If the dialstring is set up, the line will immediately be redirected out of the system not ringing any telephone.</td></tr>
<tr><td colspan="9"><b>Warning:</b> Enable modules
If you disabled any trunk media bay modules prior to performing programming, enable them now to ensure your system will function properly.</td></tr>
</table>

Restrictions (Line and Remote)

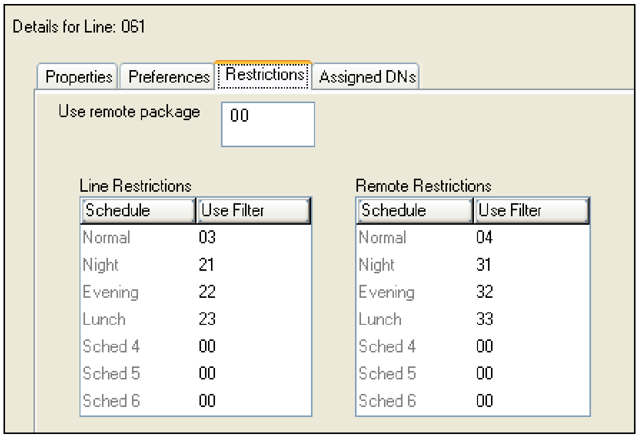
Assigning Line restrictions and Remote Access Package restrictions are part of the configuration for controlling calls out of the system (line restrictions) and into the system from a private network node or from a remote user calling in over the PSTN lines (Remote Access Packages).

The following paths indicate where to access the restriction settings in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface:****CONFIG > Terminals and Sets**

Figure 49 shows the Restrictions tab.

Figure 49 Restrictions tables for a line



Details for Line: 061

Use remote package: 00

Line Restrictions

Schedule	Use Filter
Normal	03
Night	21
Evening	22
Lunch	23
Sched 4	00
Sched 5	00
Sched 6	00

Remote Restrictions

Schedule	Use Filter
Normal	04
Night	31
Evening	32
Lunch	33
Sched 4	00
Sched 5	00
Sched 6	00

Table 42 describes the fields on this panel.

Table 42 Restrictions

Attribute	Values	Description
Use remote package	<remote package #>	If the line is being used to receive external calls or calls from other nodes on the private network, ensure that you indicate a remote package that provides only the availability that you want external callers to have. This attribute is typically used for tandeming calls.
Schedule	Default: Normal, Night, Evening, Lunch, Sched 4, Sched 5, Sched 6	
Line Restrictions - Use Filter	<00-99>	Enter the restriction filter number that applies to each schedule. (controls outgoing calls)
Remote Restrictions - Use Filter	<00-99>	Enter the restriction filter that applies to each schedule. This setting provides call controls for incoming calls over a private network or from remote user dialing in over PSTN)

Assigned DNs

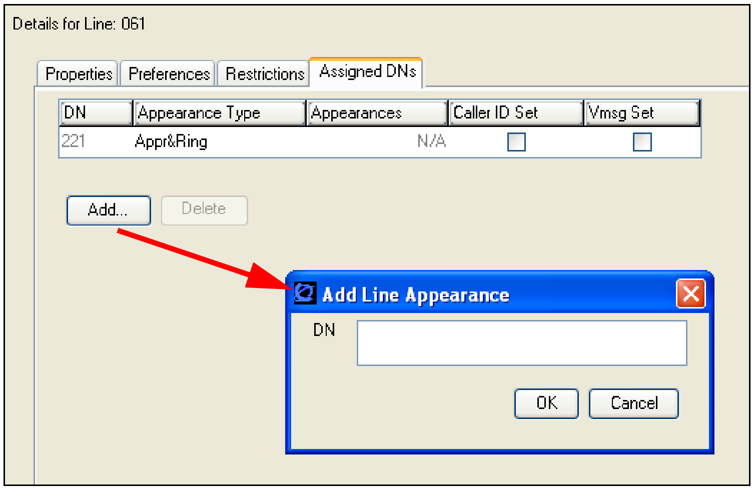
The Assigned DNs tabbed panel displays the DN properties for lines that are assigned to telephones.

This information can also be configured on the DN record. Any information added, deleted, or modified in this table reflects in the DN record.

Note: Lines that do not allow single-line assignment, such as PRI lines and VoIP lines, will not display this tabbed panel.

Figure 50 shows the Assigned DNs tab.

Figure 50 Add a DN record



DN	Appearance Type	Appearances	Caller ID Set	Vmsg Set
221	Appr&Ring	N/A	□	□

To add a DN record to a line record

1. In the top panel, click the line where you want to add a DN record.
2. In the bottom frame, click **Add**.
3. Enter the DN record number and line settings:
   - DN
   - Appearance Type
   - Appearances (target lines only)
   - Caller ID Set (for display sets and ASM8+)
   - VMsg Set
4. Click **OK**.
5. Repeat for all the DN records you want to assign.

Chapter 10
Lines overview

Telephony signals into the system, within the system, and out of the system are carried over channels. For consistency, these channels are all called lines or trunks. This designation includes:

- circuit switched lines (PSTN): connect to the system through media bay modules
- Voice over IP (VoIP) trunks: connect through the LAN or IP network
- target lines, internal channels: connect PRI, T1 and VoIP trunks to specific devices
- intercom lines: connect all internal telephones together through the DN numbers, and allow the user to access line pools for making outgoing calls, as well as being required for other call features such as conference calling and system-wide call appearance (SWCA) calls. Intercom designations are assigned in the DN record, or automatically by the system for each telephone.

Prerequisites

You must configure the media bay modules and/or the VoIP trunk parameters before you can set up line programming.

- The position on the system bus of the trunk media bay modules determines the line numbers that are available. Refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).
- The position on the system bus of the station media bay modules determines which DNs are available, although DN numbers can be changed.
- Available VoIP lines are determined by the number of VoIP keycodes entered on the system (between 01 and 60), starting with line 001 and ending at line 060.

Also refer to the following topics:

- "Understanding how the system identifies lines" on page 159
- "Line record" on page 162
- "Line Job Aids" on page 164

Other line configuration options or requirements:

- **BRI loops** require configuration and provisioning before the BRI lines can be configured.
- The BCM also offers facilities for **splitting trunks** to deliver both data and telephony services.

Understanding how the system identifies lines

On a new system, lines and loops are numbered and assigned defaults based on the type of media bay modules that have been connected to the system. The exception are the VoIP trunks, which require a keycode to activate.

These panels allow you to easily view which lines have been enabled through a media bay module.

From this heading, you can access each line record and assign attributes, as you require.

Determining which lines you need to program

Under **Lines**, note that line types are divided into five headings. The fifth heading contains all line numbers.

- Active Physical Lines
- Active VoIP Lines
- Target Lines
- Inactive Lines
- All Lines

Active physical lines

Lines 061-240 are reserved for physical lines.

Active VoIP lines (require keycode)

Voice over IP (VoIP) lines are signaling channels that simulate how CO lines work. However, VoIP lines transmit data to the IP network over a LAN or IP network rather than over physical lines. Once the VoIP trunks are set up, you can assign them to line pools, and program their behavior in the same way you would PRI lines.

VoIP lines use line numbers 001 to 060. To view these line records, select **Configuration > Telephony > Lines > Active VoIP Lines**. To access VoIP lines, you need to enter software keycodes. Each keycode supports a specific number of lines. No entries appear in the Enabled VoIP lines field until you complete the IP Trunks Settings field, which displays when you select IP Trunks under **Configuration > Resources > Telephony Resources > IP trunks**.

VoIP trunks should be configured to use a single line pool per trunk type. Do not mix other trunk types on the same line pool. The VoIP line pools are assigned to routes, which in turn, are configured with destination codes that route calls to the designated remote gateways of other BCM systems or Meridian 1-IPT systems.

You can also create a fallback for the trunk. This is a situation where the system reroutes the call to a PSTN line pool if the primary route is not available or the call quality is not suitable. If you do not configure your network for fallback and the call quality is below threshold, the IP call fails.

Target lines

Target lines are internal communications paths that directly connect auto-answer trunks to system telephones. These lines are incoming only.

Target lines allow you to make more efficient use of DID line resources.You can map a range of target lines for each DID line. The incoming call is routed according to the mapped dialed digits, rather than a one-to-one line assignment. Systems configured using the DID template automatically assign target lines to all assigned DNs.

You also require target lines when you use PRI, T1 or VoIP trunks.

Target lines use line numbers 241 to 492. To view these lines, select **Configuration > Telephony > Lines > Target Lines**. Record this information in your system Programming Records so you have a clear view of where each line is assigned.

Other features:

- Each target line can be assigned to more than one telephone.
- A telephone can have multiple appearances of a target line.

Target lines are internal direct links the BCM uses to allow external callers to dial specific system telephones, or a group of system telephones. You assign the target line to one or more telephone DNs, and then configure the target line to function as you require. You can also assign multiple appearances of a target line to one telephone. This allows more than one call to simultaneously use the target line. Target lines are required by lines that support multiple numbers over one trunk (T1 E&M, DID trunks, T1 DID trunks, PRI trunks, and VoIP trunks).

Caution: Changing the received # length:
If you change the received # length for your system, the **Public number** entry for the target lines will clear if the new received # length is less than the number entered in this field.
If the new received # length has more digits than the number entered in this field, you must change the entry manually, if changes are required.

Programming note: The following trunks use one, or both, of these settings to route calls:

- DPNSS lines use the Private received number to route calls in the system.
- BRI ETSI-QSIG, PRI ETSI-QSIG, MCDN, DMS100, DMS250 and VoIP trunks route calls on a per-call basis to either the public or private received digits.

Note: VoIP trunking MCDN calls do not support Auto DN/DISA DN functionality.

- BRI (ETSI-Euro, NI), PRI (ETSI-Euro, NI, 4ESS), T1 (LoopStart, E&M, DID, GroundStart), Analog LEC (LoopStart), and DASS2 trunks route calls using the Public received number.

Physical lines

Physical lines are the central office (CO) trunks assigned to the trunk media bay modules. Refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) for information on which lines are enabled.

You can change the line types to suit your system. For instance, BRI and DTM modules can be designated to a number of line types, depending on the type of line service provided through the central office (CO). However, the line numbers are associated for specific tasks or to specific DS30 bus numbers.

The line record allows you to program settings for lines that affect how the lines operate in the network and with other switches, as well as how the system uses the line.

Trunk types:

- VoIP
- DTM: TI types (Loop, E&M, DID, Ground, or fixed data channel), PRI, DASS2, DPNSS. The DDI MUX module contains a DTM.
- CTM (North America)/GATM: Analog Loop
- BRI: BRI S/T, BRI U2, BRI U4
- Target lines

➡ **Note:** BRI U2 and BRI U4 are only available through a FEM module connected to a Norstar trunk modules with a BRI U2 or BRI U4 card.

BRI loops programming

The Loops panels define the loop numbers and loop attributes that correspond to the DIP switch settings that were configured on the BRI trunk media bay modules installed on your system. Check your Programming Record to see which modules are installed, and what settings were chosen.

Available BRI trunk loop attributes are determined by the country profile that is assigned to your system. All profiles allow BRI programming; however, there is a difference between T1-based profiles and for E1-based profiles.

Once loops are provisioned, the system assigns two line numbers per loop. These lines are then programmed as you would any other lines.

You can program a loop to support either trunking services to the ISDN network, or terminal services to one or more ISDN devices. The following sections describe the programming for each type of loop. For complete module installation instructions and safety precautions, see the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

Programming links

Determine line assignments for routing: “Line Job Aids” on page 164.

Line record

The line record allows you to:

- Identify the line and the features on the line.
- Assign restrictions for outgoing calls.
- Assign a voice message center, if the line connects to a remote voice mail system, either on another node on the private network, or at the central office.

Line characteristics

Line type determines what features are available. Some features must be coordinated with the settings at the other end of the line.

Programming links

Alternate-click the Line Assignment panel tab to see a list of the line feature settings, and to see which lines have each setting.

Line restrictions

Restrictions prevent certain kinds of calls from occurring over specific lines. You can also restrict some features.

If you want different restrictions to apply at different times of the day or week, you can set up the line restriction schedules to that effect. The Normal schedule runs when no other schedule is specified or if fallback is used for VoIP trunks.

Table 43 lists the default restriction filters.

Table 43 Default restriction filters

Schedule	Restriction filter
Normal	03
Schedule 1 (Night)	21
Schedule 2 (Evening)	22
Schedule 3 (Lunch)	23
Schedule 4	00
Schedule 5	00
Schedule 6	00

Note: When a remote user places an external call on a line, any filters used with the line still apply.

Programming links

The template has a set of default restrictions in Restriction filter 01 only. You must create your own restriction files if you want to use other settings.

Remote restrictions

Your system can accommodate users who call in from outside the system to access system features. Calls coming in over the Private network that are routing out of the system to remote systems or to the PSTN are also considered to be remote call-ins.

To restrict the access remote callers have, or to control outbound private network calls, specify the appropriate filter for the line.

If you want different restrictions to apply at different times of the day or week, you can set up the line restriction schedules to that effect. The Normal schedule runs when no other schedule is specified or if fallback is used for VoIP trunks.

The default restrictions are shown in Table 44

Table 44 Default remote restrictions

Schedule	Restriction filter	Schedule	Restriction filter
Normal	04	Schedule 4	00
Schedule 1 (Night)	31	Schedule 5	00
Schedule 2 (Evening)	32	Schedule 6	00
Schedule 3 (Lunch)	33		

➡ **Note:** The remote restriction restricts the numbers a user can dial on an incoming auto-answer line. If a remote user selects a line to place an external call, any filter used with the line still applies.

Voice message center

If you subscribe to a voice message service outside your office, you can indicate to the line with which voice message service to connect.

Voice message centers are defined as part of the system telephony global programming. To access the voice message centers in Element Manager, select **Configuration > Applications > Voice Messaging/Contact Center**.

Line Job Aids

Refer to the following topics for additional information:

- “Determining line numbers” on page 164
- “Line pool tips” on page 168
- “Using loss packages” on page 168
- “Turn Privacy on or off for a call” on page 169

Determining line numbers

Refer to Table 45 for a list of lines assigned per bus (DS30 bus and offset), based on the module type configured with that address. You can use this chart to note which lines should be active for the modules you installed. You can also note which line pool you put the lines in, and note the line pool access codes or routes and destination codes to which you assigned the line pools (or use your programming records).

Follow these steps to use the table:

1. For each bus number, circle the module you set to that number.
2. Beside the module name, circle the group of line numbers appropriate for the offset you set on the modules.

3 In the Line pool column, indicate a line pool name if you want to associate lines into a pool. This enables assigned telephones to grab any free line from the pool.

4 On the far right column, list the access codes and routes associated with the lines.

Table 45 Line numbering for modules and VoIP (Sheet 1 of 2)

DS30 bus	**Type of module**	**Line/Loop numbers (default) Offset**				**Line pool A-O/Bloc**	**Access codes and routes**
		0	**1**	**2**	**3**		
		Default Start DN: 221					
02	Trunk modules						
	DTM (T1)	211-234					
	DTM (NA-PRI)	211-233					
	DTM (E1 PRI)	211-240					
	DDI MUX DTM	211-234*					
	BRI	211-218	219-226	227-234			
	CTM4, GATM4 and **4X16**	211-214	219-222	227-230	235-238		
	CTM8, GATM8 (upper/ lower)	211-214 219-222	219-2222 227-230	227-2302 235-238	N/A		
	ISDN loops						
	BRI ST/U2/U4	201-204	205-208	209-212			
*Note which lines are for data and which are for telephony.							
03	Trunk module						
	DTM (T1)	181-204					
	DTM (NA-PRI)	181-203					
	DTM (E1 PRI)	181-210					
	DDI MUX DTM	181-204*					
	BRI	181-188	189-196	197-204			
	CTM4, GATM4 and **4X16**	181-184	189-192	197-200	205-208		
	CTM8, GATM8 (upper/ lower)	181-184 189-192	189-192 197-200	197-200 205-208	N/A		
	ISDN loops						
	BRI ST/U2/U4	301-304	305-308	309-312			
*Note which lines are for data, and which are for telephony.							
04	Trunk module						
	DTM (T1)	151-174					
	DTM (NA-PRI)	151-173					
	DTM (E1 PRI)	151-180					
	DDI MUX DTM	151-174*					
	BRI	181-188	189-196	197-204			

Table 45 Line numbering for modules and VoIP (Sheet 2 of 2)

DS30 bus	Type of module	Line/Loop numbers (default) Offset				Line pool A-O/Bloc	Access codes and routes
		0	1	2	3		
04 (cont)	**CTM4, GATM4** and **4X16**	151-154	159-162	167-170	175-178		
	CTM8, GATM8 (upper/ lower)	151-154 159-162	159-162 167-170	167-170 175-178	N/A		
	ISDN loops						
	BRI ST/U2/U4	401-404	405-408	409-412			
*Note which lines are for data, and which are for telephony.							
05	Trunk module						
	DTM (T1)	121-144					
	DTM (NA-PRI)	121-143					
	DTM (E1 PRI)	121-150					
	DDI MUX DTM	121-144*					
	BRI	151-158	159-166	167-174			
	CTM4, GATM4 and **4X16**	121-124	129-132	137-140	145-148		
	CTM8, GATM8 (upper/ lower)	121-124 129-132	129-132 137-140	137-140 145-148	N/A		
	ISDN loops						
	BRI ST/U2/U4	501-504	505-508	509-512			
*Note which lines are for data, and which are for telephony.							
06	Trunk module						
	DTM (T1)	91-114					
	DTM (NA-PRI)	91-113					
	DTM (E1 PRI)	91-120					
	DDI MUX DTM	91-114*					
	BRI	91-98	99-106	107-114			
	CTM4, GATM4 and **4X16**	91-94	99-102	107-110	115-188		
	CTM8, GATM8 (upper/ lower)	91-94 99-102	99-102 107-110	107-110 115-188	N/A		
	ISDN loops						
	BRI ST/U2/U4	601-604					
*Note which lines are for data, and which are for telephony.							

Line pool tips

Line pools are groups of lines. Pooling lines allows you to use fewer lines than there are users. PRI lines and VoIP lines are always defined into line pools.

- Line pools must never contain a mixture of lines. All lines in a given line pool should go to the same location.
- Avoid putting unsupervised loop start lines in a line pool. These lines can become unusable, especially when a remote user uses the line pool to make an external call.
- Assign line pool access to telephones (select **Configuration > Telephony > Dialing Plan > Line Pools**.)
- Assign system-wide line pool access codes (select **Configuration > Telephony > Dialing Plan > General)** (not applicable to Bloc pools).
- A telephone can be administered to search automatically for an idle line from several lines that appear on the telephone. Assign a line pool as the prime line. When the user lifts the receiver or presses Handsfree, any one of the lines, if idle, can be selected by Automatic Outgoing Line selection.
- Changes in the settings for trunk type on a system that is in use can result in dropped calls.
- When assigning lines to line pools, consider your network configuration. You can create a unified dialing plan by assigning lines to the same location to the same line pool on each of your systems. For example, if system A and system B each have TIE lines to system C, assign the TIE lines to pool D on each of the systems. You cannot assign target lines to a line pool, as they are incoming-only.

Using loss packages

The loss package settings allow you to select the appropriate loss/gain and impedance settings for each line. The setting is based on the terminating switch type and the distance between BCM and the terminating switch.

When measuring the distance from BCM to CO and from BCM to PBX systems, use 600 ohms as the termination resistance setting.

Table 46 Loss package settings

Loss Package	Receive Loss	Transmit Loss	Impedance	Distance to switch/cable loss/terminating switch
Short CO	0 dB	3 dB	Short	Short/<2 dB/BCM to CO
Medium CO	0 dB	0 dB	TIA/EIA 464	Medium/>2 dB and <6 dB/BCM to CO
Long CO	-3 dB	0 dB	TIA/EIA 464	Long/>6 dB/BCM to CO
Short PBX	0 dB	0 dB	Short	Short/<2 dB/BCM to PBX
Long PBX	-3 dB	0 dB	TIA/EIA 464	Long/>2 dB/BCM to PBX

A loss of 4 dB corresponds to a cable length of approximately 2700 m (9000 ft).

Note: Loss packages are not supported on the 4X16 combo.

Turn Privacy on or off for a call

You can configure lines in your system to have automatic privacy. With a line not programmed with privacy, anyone with the line assigned to their telephone can join your call by pressing the line button. With a line programmed with privacy, one person at a time can use the line.

Use **FEATURE 83** to turn the Privacy feature off and on.

Privacy control cannot be used for internal or conference calls.

When another telephone joins a call, the participants on the call hear a tone, and a message appears on the telephone display. It is not possible to join a call without everyone hearing this tone.

➡ **Note:** The Auto privacy setting does not apply to target lines, PRI lines or VoIP trunking lines.

Programming line access

There are a number of ways you can configure your lines. You can assign each line to one telephone or several telephones, or a specific line to a specific telephone. You can also pool your lines so that a number of telephones have access to several lines.

Refer to the following topics:

- "Making lines available"
- "Incoming calls" on page 170
- "Outgoing calls" on page 170

Making lines available

- You can determine whether a line will be assigned solely to one telephone, or if a group of users will have access to the line.
- Even when you use line pools, it is possible that a line pool will be unavailable for outgoing traffic. To alleviate this, you can determine overflow paths for any routes that you designate.
- Incoming lines can be assigned to telephones as individual lines or through target lines, depending on the type of trunk supplied from the central office (CO). Incoming lines do not need to have an appearance on the telephone. Target lines are for incoming calls only. Two-way single lines, such as analog lines, allow the user to make an outgoing call by pressing the (idle) assigned line button or, if the line is part of a line pool, by entering a line pool access code or destination code to access the line pool. These lines can also be redirected on a per-trunk basis through Element Manager or from the telephone by using **FEATURE 84**.
- PRI lines are always configured into line pools. These lines require a destination code for outgoing calls. Incoming calls use target line assignments.
- Voice over IP (VoIP) trunks use the data network to provide line service in and out of the system. VoIP trunk configuration is described in the*BCM 4.0 Device Configuration Guide* (N0060600). VoIP trunks use target lines for incoming calls, and require line pool codes or destination codes for outgoing calls.

- You can assign a line a maximum of 93 times.

Incoming calls

For incoming calls, you can have a central answering position, or you can specify lines to one or more telephones to receive directed calling.

You can arrange your telephones in Hunt groups, ringing groups, or call groups that use system-wide call appearance (SWCA) assignments to share calls.

You can also configure lines for use by system users who call in from outside the system. You can give them direct access to the system with an Auto DN, or you can configure the line so they hear a stuttered dial tone, at which point they need to enter a password (CoS) to gain access (DISA DN).

Outgoing calls

For outgoing calls, you can assign one or more intercom keys to access a line pool or prime line, destination code, or internal system numbers to direct the call. Telephones without intercom keys do require intercom paths assigned, but to access calls, users must pick up the handset to connect.

For calls within the system, all telephones are virtually linked within the system. To call another telephone inside the system, you can lift the handset and dial the local DN. In this case, the prime line must be set to intercom.

For calls going outside the system:

- If you assign the prime line to a line pool — When you pick up the handset, the telephone automatically grabs the first available line from the assigned line pool. In this configuration, you must ensure that the outgoing number is allowed by the line pool.
- If you assign the prime line to an intercom button — You can enter a line pool access code or a destination code followed by the telephone number to direct the outgoing call where it exits the system on any available line in that pool.

Chapter 11
Configuring lines: PRI

PRI are auto-answer lines. These lines cannot be individually assigned to telephones. They must be configured into line pools. PRI line pools then are assigned routes and these routes are used to create destination codes.

The following paths indicate where to access PRI line pools in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Lines**

Task: Configure the PRI lines connected to the system

- "Configuring PRI line features" on page 173
- "Configuring PRI Call-by-Call services" on page 174

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

Install and configure the DTM module. Refer to "Trunk Module Parameters" on page 122.	
Provision lines. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 51 and Figure 52 provide an overview of the PRI line feature configuration process.

Figure 51 PRI line feature configuration process — Part A

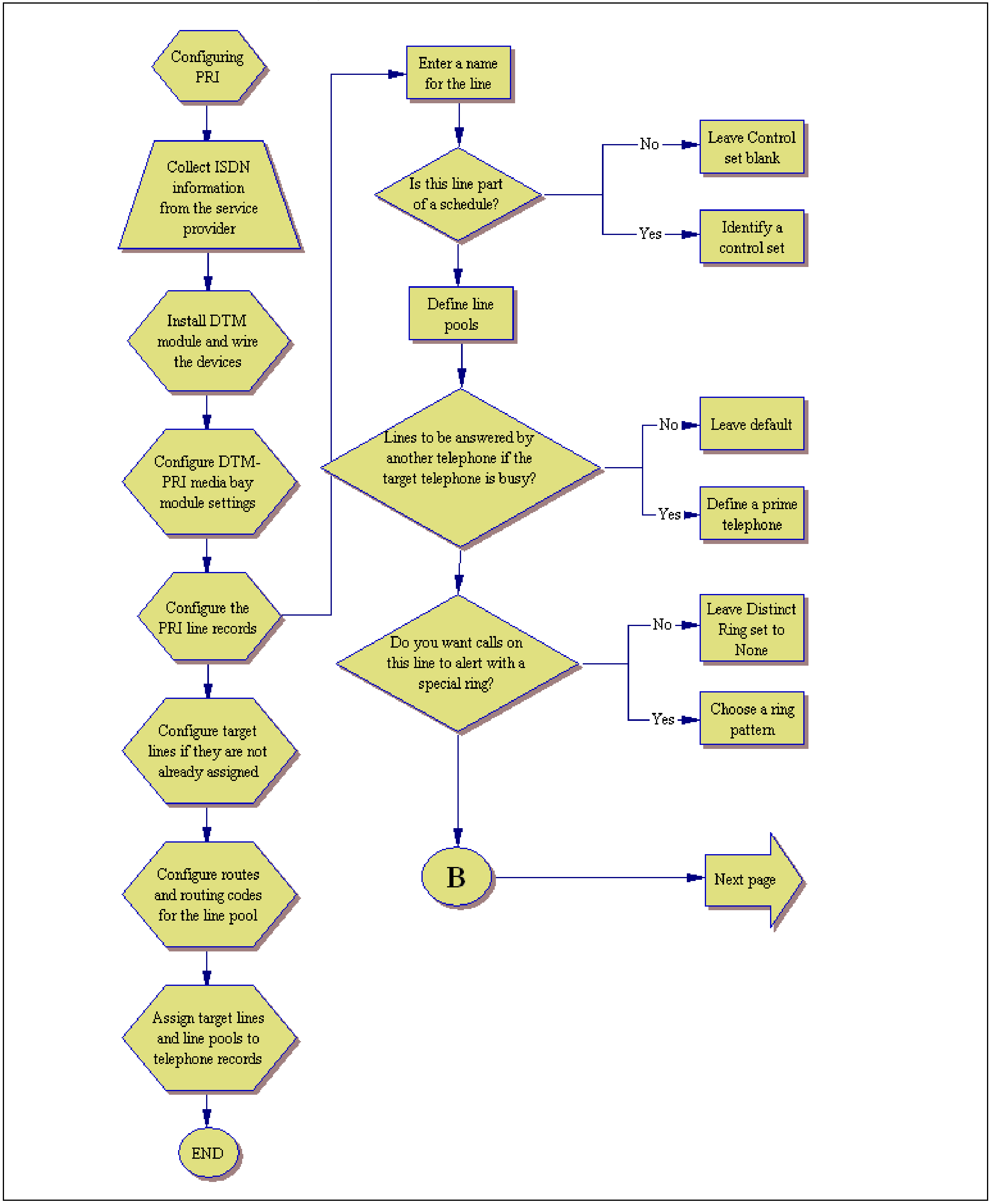
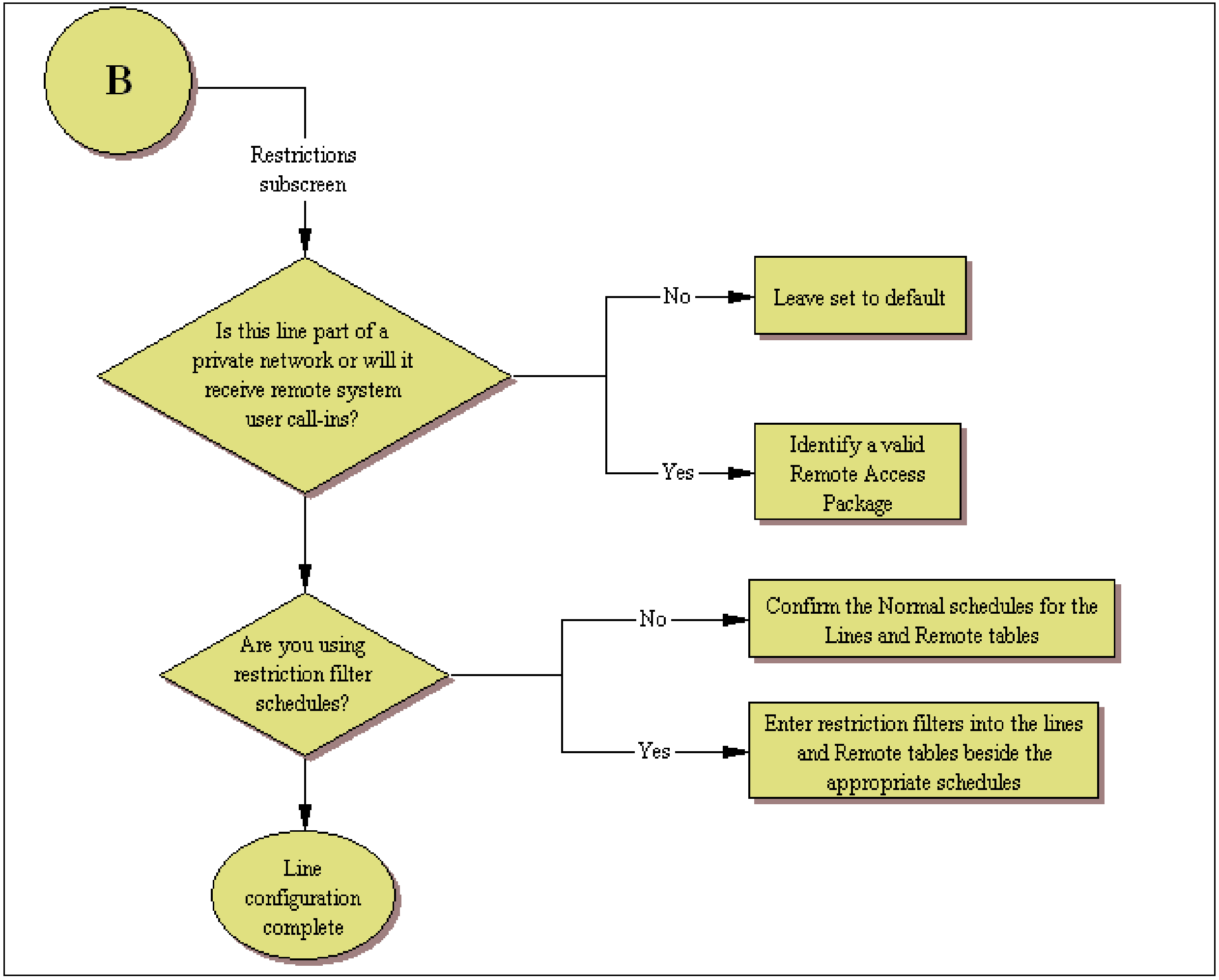

Configuring PRI
Collect ISDN information from the service provider
Install DTM module and wire the devices
Configure DTM-PRI media bay module settings
Configure the PRI line records
Configure target lines if they are not already assigned
Configure routes and routing codes for the line pool
Assign target lines and line pools to telephone records
END
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set blank
Yes
Identify a control set
Define line pools
Lines to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
B
Next page

Figure 52 PRI line feature configuration process — Part B


B
Restrictions subscreen
Is this line part of a private network or will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedules for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Line configuration complete

Configuring PRI line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Line: Number of the line being assigned.
- Trunk Type: PRI or ETSI (European standard).
- Name: Identify the line or line function.
- Control Set: Identify a DN if you are using this line with scheduling.
- Line Type: Define how the line will be used. If you are using routing, ensure it is put into line pool (BlocA to BlocF). If you use line pools, you need to assign target lines to the telephones, as well (refer to "Configuring lines: Target lines" on page 177).
- Prime Set: If you want the line to be answered at another telephone, if the line is not answered at the target telephone.

- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).

Subpanel under Restrictions tab:

- Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

These lines cannot be assigned to DNs as line assignments. They are assigned only as line pools. Instead, configure target lines for each telephone and assign the target line to the telephones. For more information, refer to the *BCM 4.0 Device Configuration Guide* (N0060600).

2 Suggested next steps:

- Dialing plan sections
  "Dialing plan: System settings" on page 303
  "Dialing plan: Public network" on page 311
  "Dialing plan: Private network settings" on page 317
  "Dialing plan: Routing and destination codes" on page 289
- Networking sections
  "Public networking: Tandem calls from private node" on page 327
  "Private networking: Using destination codes" on page 373
  "Private networking: PRI Call-by-Call services" on page 377
  "Private networking: PRI and VoIP tandem networks" on page 357
  "Private networking: MCDN and ETSI network features" on page 353
  "Private networking: MCDN over PRI and VoIP" on page 329

Configuring PRI Call-by-Call services

Call-by-Call service selection (CbC) allows you to access services or private facilities over a PRI line without the need for dedicated facilities. The different services represent different types of access to the network.

The following protocols support Call-by-Call limits:

- National ISDN 2 (NI-2)
- DMS100 custom
- DMS250
- AT&T 4ESS custom

There are several areas in the interface where you need to configure Call-by-Call services and the PRI lines that support these services.

To configure Call-by-Call services and the PRI lines

1. Set up the DTM module to support PRI.
2. Set up the Call-by-Call services selection for the module. Refer to "Call-by-Call Service Selection" on page 126.
3. Provision the PRI lines. Refer to "Provisioning module lines/loops" on page 130.
4. Configure the PRI lines. Refer to "Configuring lines: PRI" on page 171.
5. Configure target lines, if they are not already configured for your system. Refer to "Configuring lines: Target lines" on page 177.
6. Assign the PRI line pools to telephones. Refer to "Line Access - Line Pool Access tab in the *BCM 4.0 Device Configuration Guide* (N0060600).
7. Assign the target lines to telephones. Refer to "Line Access - Line Pool Access tab in the *BCM 4.0 Device Configuration Guide* (N0060600) and Line pools: DNs tab in the *BCM 4.0 Device Configuration Guide* (N0060600).
8. Set up routing for the PRI pools. Refer to "Programming the PRI routing table" on page 285.
9. Set up call-by-call limits for the line pools. Refer to "Line pools: Call-by-Call Limits tab (PRI only)" on page 300. Set up routing scheduling for the PRI line pools.

Chapter 12

Configuring lines: Target lines

Target lines are virtual lines that allow the mapping of received digits to a line number.

The following paths indicate where to access target lines in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines > Target Lines**
- Telset interface: ****CONFIG > Lines**

Task: Configure Target lines and DASS2 line settings
• “Configuring Target line settings” on page 180

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

Ensure that external number is mapped to internal received number, if required.	
Have a list of DNs where the target lines will get assigned.	
For features that require target lines: • Configure lines into line pools. Refer to “Trunk Module Parameters” on page 122. • Routing and destination codes. Refer to “Dialing plan: Routing and destination codes” on page 289. • Set up VoIP fallback. Refer to “Setting up VoIP trunks for fallback” on page 423.	

Process map

Figure 53 and Figure 54 provide an overview of the target line feature configuration process.

Figure 53 Configuring target lines — main screen

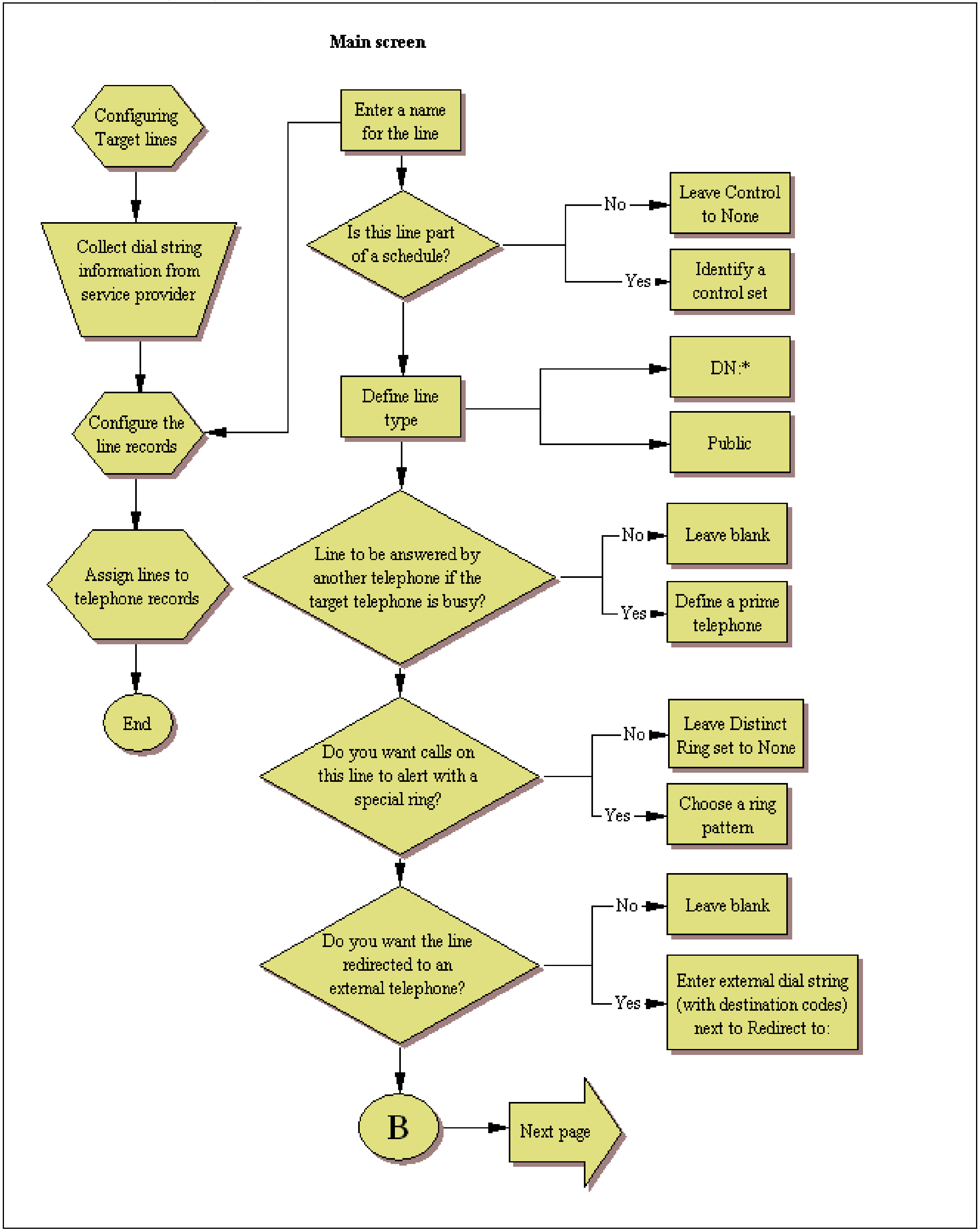
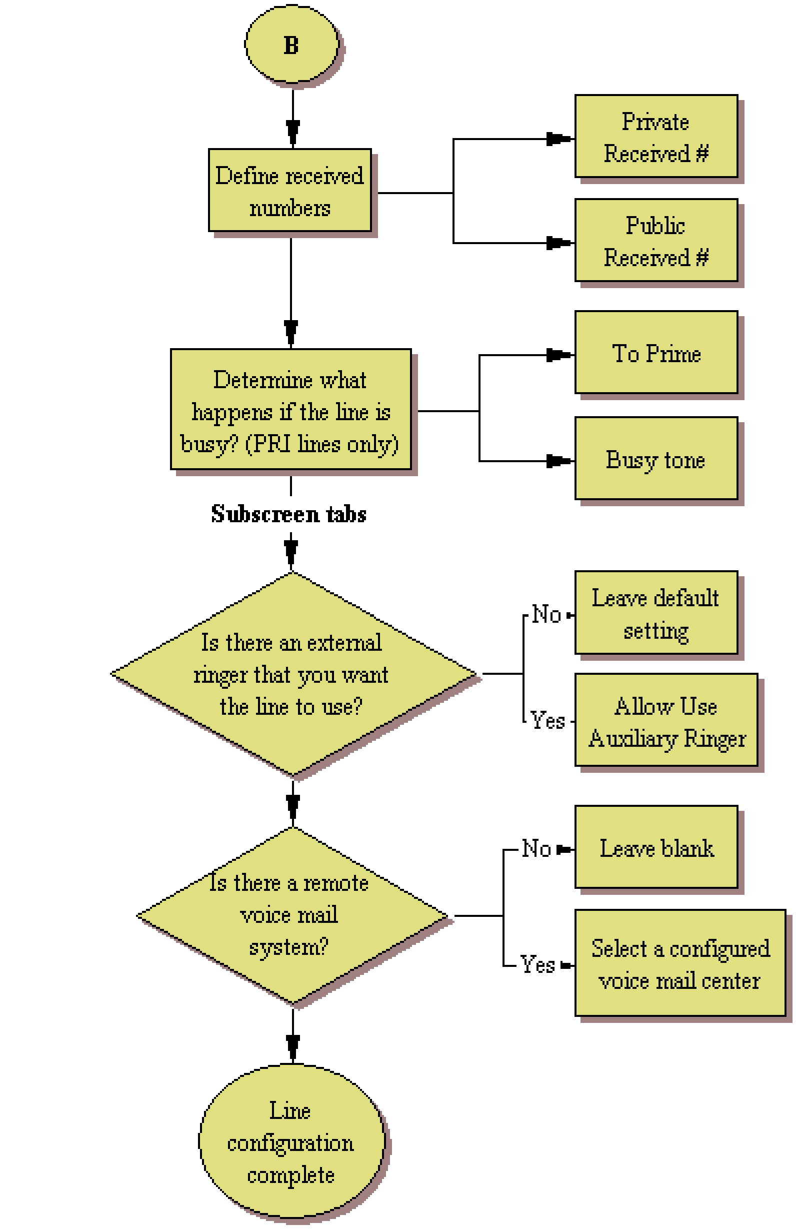

Main screen
Configuring Target lines
Collect dial string information from service provider
Configure the line records
Assign lines to telephone records
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control to None
Yes
Identify a control set
Define line type
DN:*
Public
Line to be answered by another telephone if the target telephone is busy?
No
Leave blank
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
Do you want the line redirected to an external telephone?
No
Leave blank
Yes
Enter external dial string (with destination codes) next to Redirect to:
B
Next page

Figure 54 Configuring target lines — subscreens


B
Define received numbers
Private Received #
Public Received #
Determine what happens if the line is busy? (PRI lines only)
To Prime
Busy tone
Subscreen tabs
Is there an external ringer that you want the line to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Is there a remote voice mail system?
No
Leave blank
Yes
Select a configured voice mail center
Line configuration complete

Configuring Target line settings

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Line: Number of the assigned line.
- Trunk Type: Target line.
- Name: Identify the line or line function.
- Control Set: Identify a DN if you are using this line with scheduling.
- Line Type: Set to Public, if the line is to be shared among telephones or DN:*: if the line is only assigned to one telephone.
- Prime Set: If the line is to be answered at another telephone if the line is not answered at the target telephone.
- Pub. Received #: Confirm the existing number or enter a public received # (PSTN DID or PRI trunks) that the system will recognize as the target telephone/group.
- Private Received #: If private network trunks (PRI or VoIP trunks) are configured, enter a Private received #. This number is usually the same as the DN.
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).

2 Configure the trunk/line data (Preferences tab):

- Aux. ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- If Busy: To automatically direct calls to the prime telephone, select To prime from the drop-down menu, or select Busy tone.
- Voice message center: If the system is using a remote voice mail, pick the center configured with the contact number.
- Redirect to: To automatically direct calls out of the system to a specific telephone, such as a headoffice answer attendant, enter that remote number here. Ensure that you include the proper routing information.

3 Assign the lines to DNs (see "Assigned DNs" on page 157):

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs here. The DN record can also be used to assign lines; refer to "Line Access - Line Assignment tab in the *BCM 4.0 Device Configuration Guide* (N0060600).

- DN: Unique number.
- Appearance Type: Choose Appr only or Appr&Ring if the telephone has an available button, otherwise choose Ring only.
- Appearances: Target lines can have more than one appearance, so that multiple calls can be accommodated. For telephones that have these lines set to Ring only, set to None.
- Caller ID Set: Select check box to display caller ID for calls coming in over the target line.
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

Chapter 13

Configuring lines: T1-E&M

The following paths indicate where to access the E&M lines in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Lines**

Task: Configure T1 E&M lines connected to the system

- "Configuring E&M line features" on page 185

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

DTM module: Installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 55, Figure 56, and Figure 57 provide an overview for configuring the line features for T1-E&M lines.

Figure 55 T1-E&M line configuration process — Part A

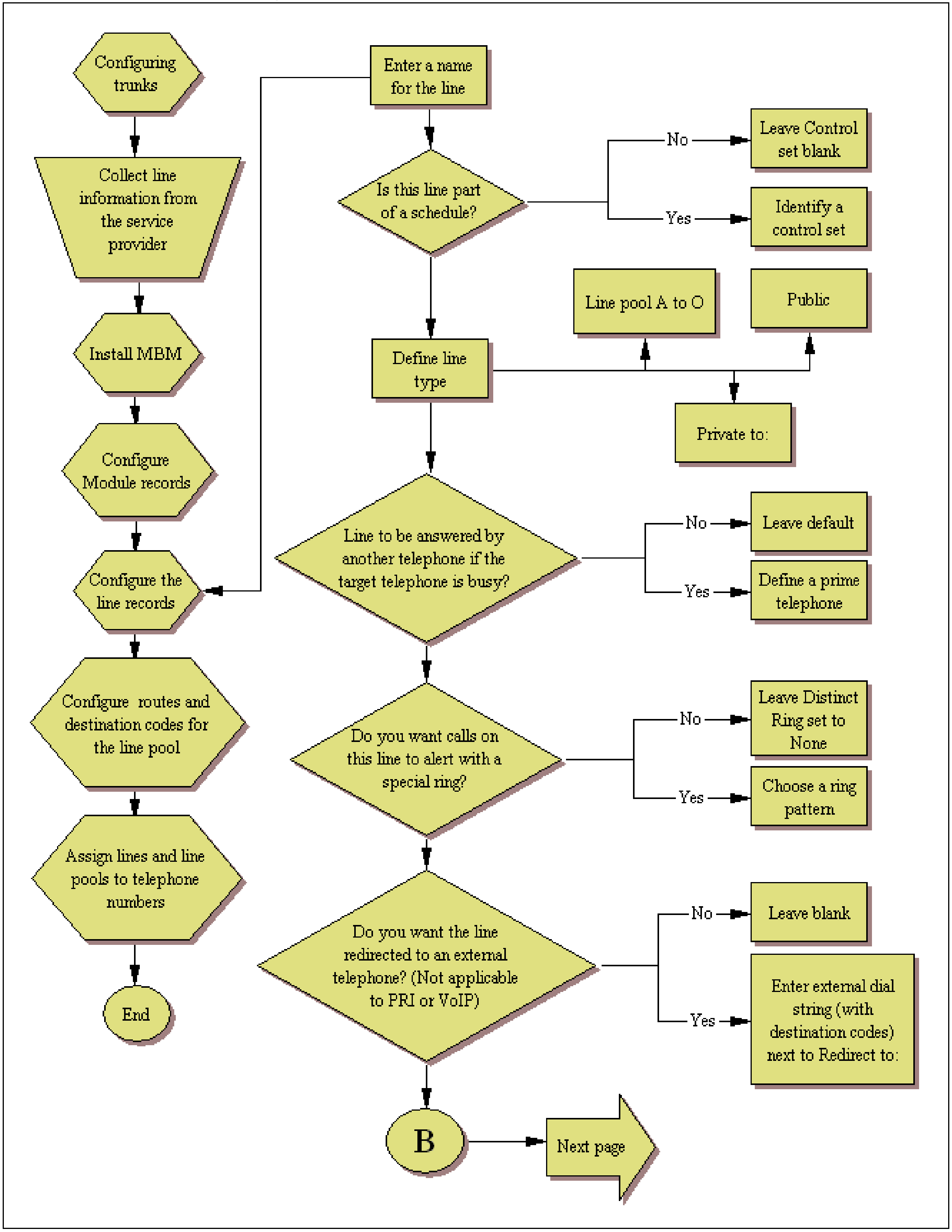

Configuring trunks
Collect line information from the service provider
Install MBM
Configure Module records
Configure the line records
Configure routes and destination codes for the line pool
Assign lines and line pools to telephone numbers
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set blank
Yes
Identify a control set
Define line type
Line pool A to O
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
Do you want the line redirected to an external telephone? (Not applicable to PRI or VoIP)
No
Leave blank
Yes
Enter external dial string (with destination codes) next to Redirect to:
B
Next page

Figure 56 T1-E&M line configuration process — Part B

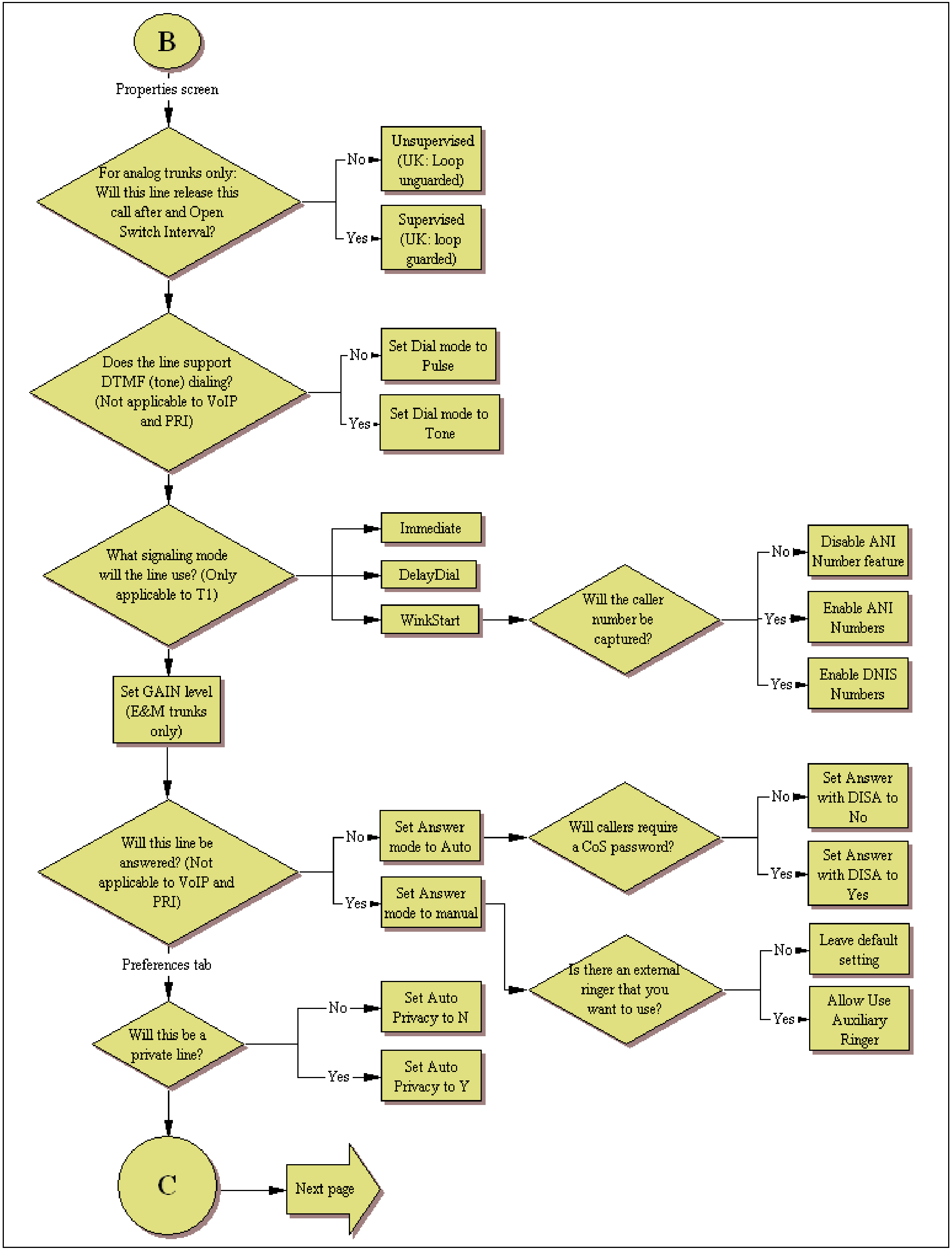
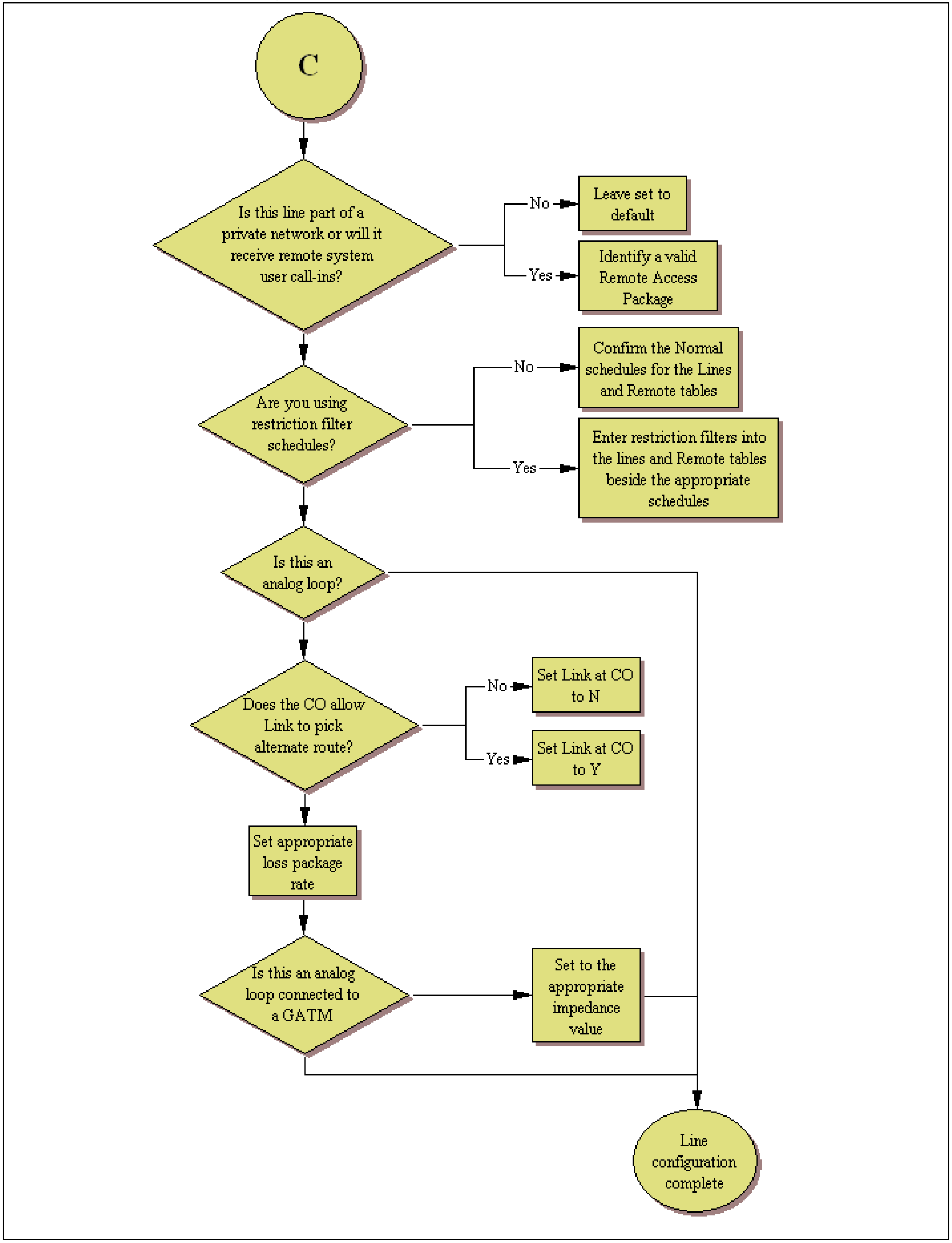

B
Properties screen
For analog trunks only: Will this line release this call after and Open Switch Interval?
No
Unsupervised (UK: Loop unguarded)
Yes
Supervised (UK: loop guarded)
Does the line support DTMF (tone) dialing? (Not applicable to VoIP and PRI)
No
Set Dial mode to Pulse
Yes
Set Dial mode to Tone
What signaling mode will the line use? (Only applicable to T1)
Immediate
DelayDial
WinkStart
Will the caller number be captured?
No
Disable ANI Number feature
Yes
Enable ANI Numbers
Yes
Enable DNIS Numbers
Set GAIN level (E&M trunks only)
Will this line be answered? (Not applicable to VoIP and PRI)
No
Set Answer mode to Auto
Will callers require a CoS password?
No
Set Answer with DISA to No
Yes
Set Answer with DISA to Yes
Yes
Set Answer mode to manual
Is there an external ringer that you want to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Preferences tab
Will this be a private line?
No
Set Auto Privacy to N
Yes
Set Auto Privacy to Y
C
Next page

Figure 57 T1-E&M line configuration process — Part C


C
Is this line part of a private network or will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedules for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Is this an analog loop?
Does the CO allow Link to pick alternate route?
No
Set Link at CO to N
Yes
Set Link at CO to Y
Set appropriate loss package rate
Is this an analog loop connected to a GATM
Set to the appropriate impedance value
Line configuration complete

Configuring E&M line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Line: Line number.
- Trunk Type: E&M.
- Name: Identify the line or line function.
- Control Set: Identify a DN if you are using this line with scheduling.
- Line Type: Define how the line will be used. If you are using routing, ensure it is put into line pool (A to O). If you use line pools, you also need to configure target lines and assign the target lines to DNs. Refer to "Configuring Target line settings" on page 180.
- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, or 4).
- Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

2 Configure the trunk/line data (Properties tab):

- Dial mode: The line service will dictate whether this needs to be set to Pulse or Tone (DTMF) dialing. These are the only two sections available
- Signaling: Match this choice with the information supplied by the service provider.

3 Set the preferences (Preferences tab):

- Auto privacy: If you activate this feature, the line is available only to the telephone that answers the call.
- Aux. ringer: Use if your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- ANI number: Enable if the caller number is to be logged. For T1 lines, this only appears if Signaling is set to WinkStart.
- DNIS number: Defines whether the digits dialed by an external caller on this line will be shown.
- Answer mode: If this line is used for remote call-ins, determine how you want the line to answer (Auto or Manual). If the answer mode is set to Auto, decide whether the caller will be immediately connected to the system or whether a stuttered dial tone will require the caller to enter a CoS password.
- Voice message center: If the system is using a remote voice mail, pick the center configured with the contact number.
- Distinct rings: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).

- Redirect to: If you want to automatically direct calls out of the system to a specific telephone, such as a headoffice answer attendant, enter that remote number here. Ensure that you include the proper routing information.

4 Set the restriction and remote package scheduling (Restrictions tab):

- Use remote package: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)
- Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Restrictions: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

5 Assign the lines to DNs (Assigned DNs tab)(applicable to manual answer only)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs, here. The DN record also can be used to assign lines and line pools for these lines.

- DN: Unique number.
- Appearance type: Choose Appr or Appr&ring if the telephone has an available button, otherwise choose Ring only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- Vmsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

6 Suggested next steps:

- Dialing plan sections
  “Dialing plan: System settings” on page 303
  “Dialing plan: Public network” on page 311
  “Dialing plan: Routing and destination codes” on page 289)
- Networking sections
  “Public networking: Setting up basic systems” on page 323
  “Public networking: Tandem calls from private node” on page 327
  “Private networking: Using shared line pools” on page 347
  “Private networking: Using destination codes” on page 373

Chapter 14
Configuring lines: T1-Loop start

Loop start trunks provide remote access to the BCM from the public network. They must be configured to auto-answer to provide remote system access.

The following paths indicate where to access the loop start trunks information through Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Lines**

Task: Configure the analog or digital loop start lines connected to the system.
• "Configuring digital (T1/E1) loop start lines" on page 191

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

Analog or DTM module is installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 58, Figure 59, and Figure 60 provide an overview of the configuration process for T1-Loop start lines.

Figure 58 T1-Loop start line configuration process — Part A

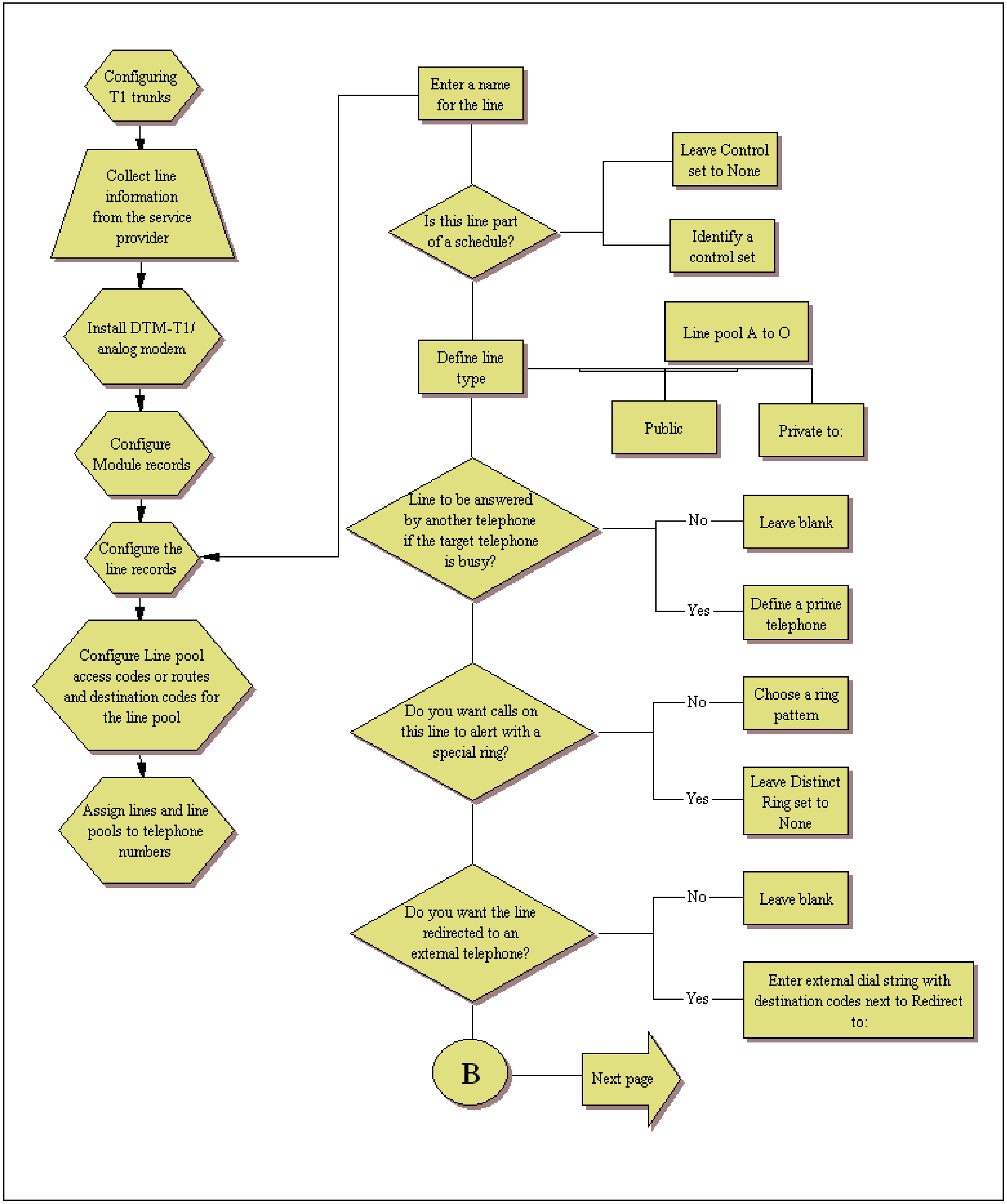

Configuring T1 trunks
Collect line information from the service provider
Install DTM-T1/ analog modem
Configure Module records
Configure the line records
Configure Line pool access codes or routes and destination codes for the line pool
Assign lines and line pools to telephone numbers
Enter a name for the line
Is this line part of a schedule?
Leave Control set to None
Identify a control set
Define line type
Line pool A to O
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave blank
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Choose a ring pattern
Yes
Leave Distinct Ring set to None
Do you want the line redirected to an external telephone?
No
Leave blank
Yes
Enter external dial string with destination codes next to Redirect to:
B
Next page

Figure 59 T1-Loop start line configuration process — Part B

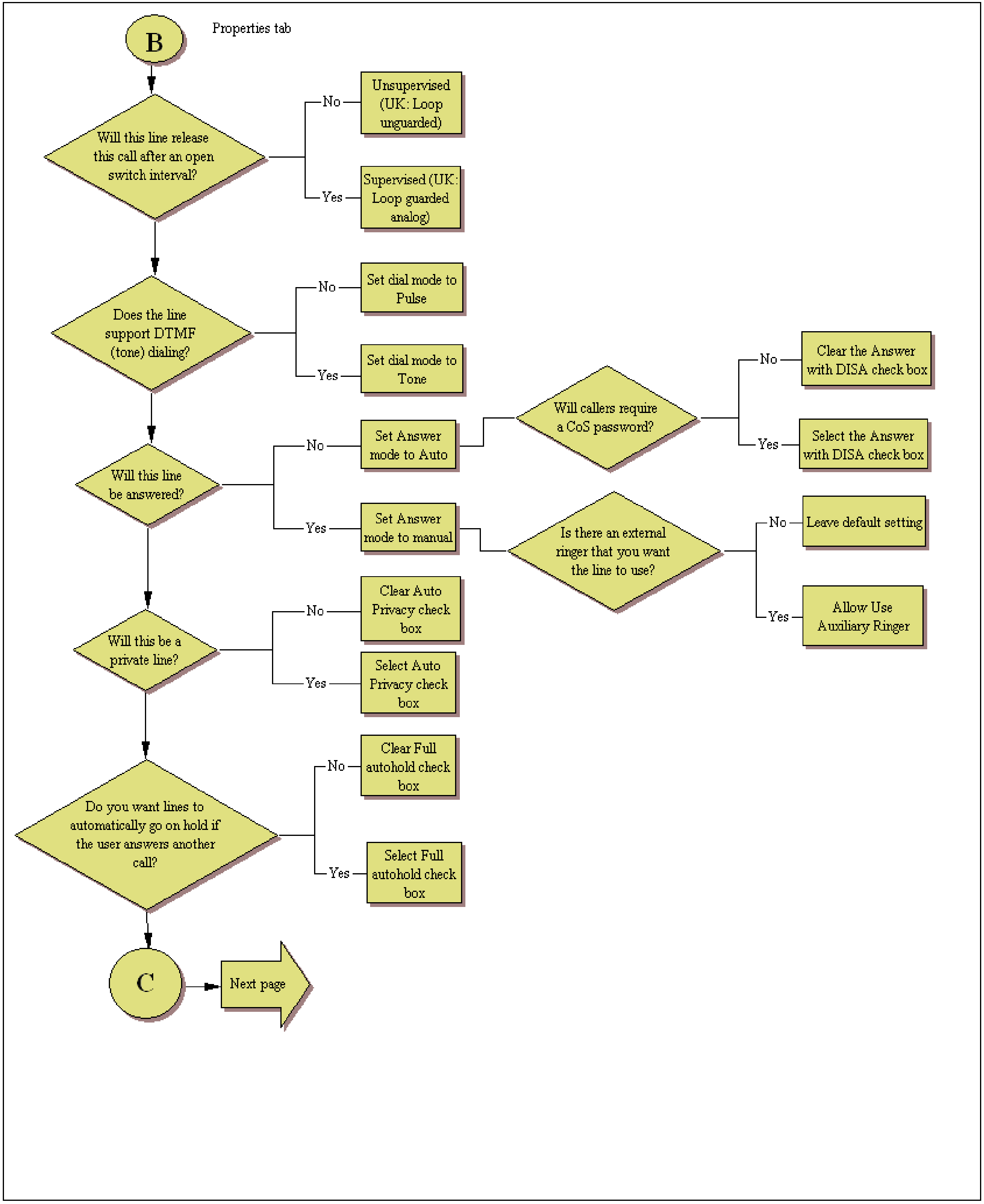
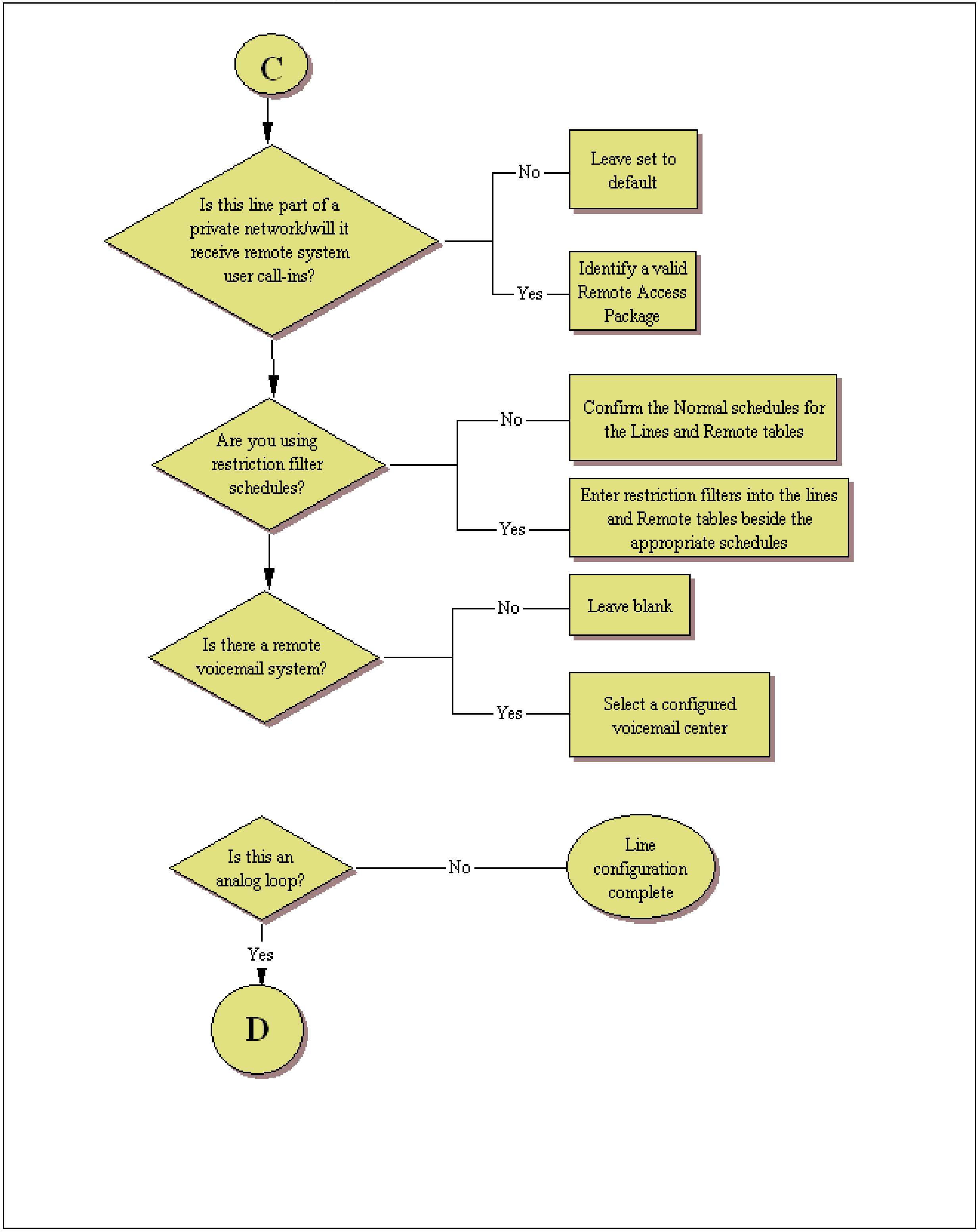

B
Properties tab
Will this line release this call after an open switch interval?
No
Unsupervised (UK: Loop unguarded)
Yes
Supervised (UK: Loop guarded analog)
Does the line support DTMF (tone) dialing?
No
Set dial mode to Pulse
Yes
Set dial mode to Tone
Will this line be answered?
No
Set Answer mode to Auto
Will callers require a CoS password?
No
Clear the Answer with DISA check box
Yes
Select the Answer with DISA check box
Yes
Set Answer mode to manual
Is there an external ringer that you want the line to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Will this be a private line?
No
Clear Auto Privacy check box
Yes
Select Auto Privacy check box
Do you want lines to automatically go on hold if the user answers another call?
No
Clear Full autohold check box
Yes
Select Full autohold check box
C
Next page

Figure 60 T1-Loop start line configuration process — Part C


C
Is this line part of a private network/will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedules for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Is there a remote voicemail system?
No
Leave blank
Yes
Select a configured voicemail center
Is this an analog loop?
No
Line configuration complete
Yes
D

Configuring digital (T1/E1) loop start lines

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to “Configuring lines” on page 147.

To configure digital loop start lines

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Line: Read-only list shows available lines for system.
- Trunk Type: Loop.
- Control Set: If you use schedules, enter DN for telephone that controls line schedules.
- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.

Under the Parameters tab:

- Name: Default name is line number, shown as part of incoming CLID.
- Line Type: Define as public, if the line is shared or as Private To (DN) if the line is assigned to a specific telephone, or put it in a line pool (A to O).
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).

Under the Properties tab:

- Trunk mode: Define whether the line will detect the open switch interval (OSI) when a call is released (supervised). Note: UK profiles use Loop guarded/Loop unguarded.
- Dial mode: The line service will dictate whether this needs to be set to Pulse or Tone (DTMF) dialing.

2 Configure the trunk/line data (Preferences tab):

- Auto privacy: If you activate this feature, the line is available only to the telephone that answers the call.
- Full autohold: This allows telephones to put a line on hold if the user picks up another line or starts to dial out on another line.
- Aux. ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- Distinct rings in use: Indicates if a special ring has been assigned.
- Answer mode/Answer with DISA: If this line is used for remote call-ins, determine how you want the line to answer (automatically, or requiring more user input). If the answer mode is set to Auto, decide whether the caller will be immediately connected to the system or whether a stuttered dialtone will require the caller to enter a CoS password.
- Voice Message Center: If the system is using a remote voice mail, pick the center configured with the contact number.

- Redirect to: If you want to automatically direct calls out of the system to a specific telephone, such as a headoffice answer attendant, enter that remote number here. Ensure that you include the proper routing information.

Under the Restrictions tab:

- Use remote package: If this line allows remote call-ins, ensure that you define a remote package.
- Line Restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Restrictions: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

3 Assign the lines to DNs (“Assigned DNs” on page 157)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs here. The DN record also can be used to assign lines.

- DN: Unique number
- Appearance Type: Choose Appr only or Appr&Ring if the telephone has an available button with indicator, otherwise choose Ring only. The 7000 and 7100 digital phones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

4 If the lines are assigned to a line pool:

- assign the line pool to DNs (“Line Access - Line Pool Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600))
- also assign a target line to the DN record. (“Line Access - Line Assignment tab” and,“Line Access - Line Pool Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).)

5 Suggested next steps:

- Dialing plan sections
  “Dialing plan: System settings” on page 303
  “Dialing plan: Public network” on page 311
  “Dialing plan: Routing and destination codes” on page 289
- Networking sections
  “Public networking: Setting up basic systems” on page 323
  “Public networking: Tandem calls from private node” on page 327
  “Private networking: Using shared line pools” on page 347
  “Private networking: Using destination codes” on page 373

Chapter 15

Configuring lines: T1-Digital Ground Start

The following describes how to configure T1-Digital Ground Start lines.

The following paths indicate where to access the Ground Start lines through Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Lines**

Task: Configure ground start lines connected to the system
• "Configuring digital ground start line features" on page 196

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

DTM module is installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 61 and Figure 62 provide an overview of the line features for Ground Start lines.

Figure 61 T1-Digital Ground Start lines configuration process — Part A

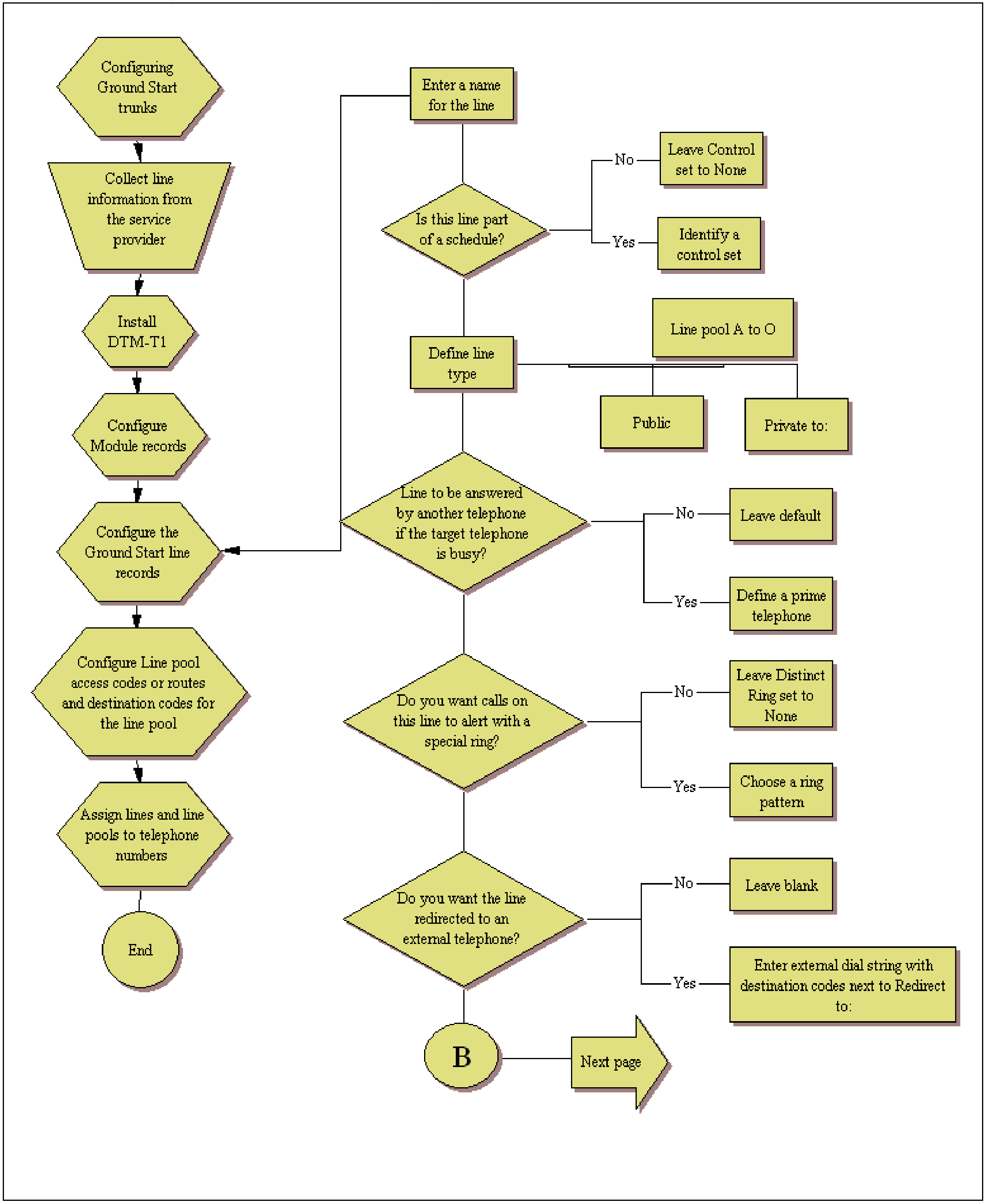
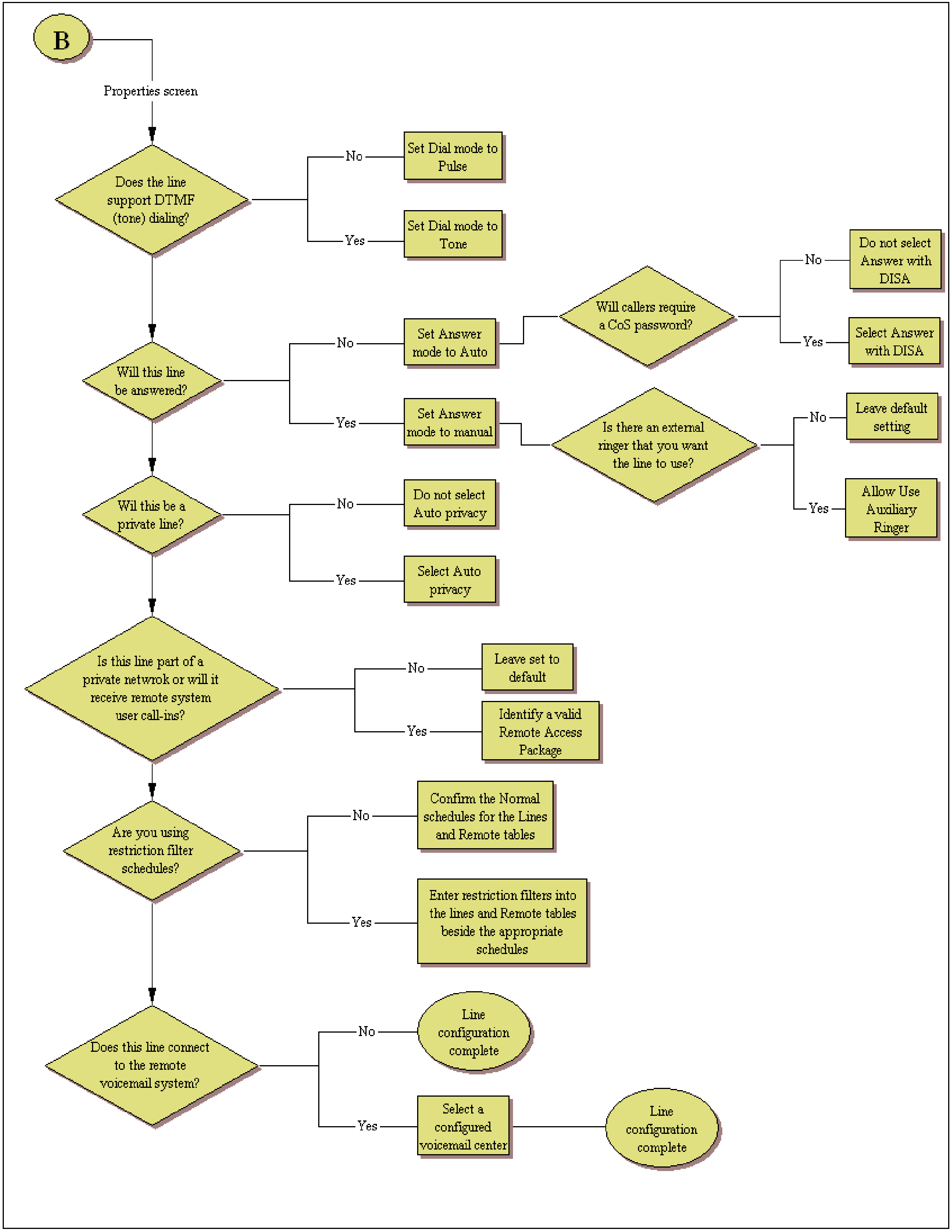

Configuring Ground Start trunks
Collect line information from the service provider
Install DTM-T1
Configure Module records
Configure the Ground Start line records
Configure Line pool access codes or routes and destination codes for the line pool
Assign lines and line pools to telephone numbers
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set to None
Yes
Identify a control set
Define line type
Line pool A to O
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
Do you want the line redirected to an external telephone?
No
Leave blank
Yes
Enter external dial string with destination codes next to Redirect to:
B
Next page

Figure 62 T1-Digital Ground Start lines configuration process — Part B


B
Properties screen
Does the line support DTMF (tone) dialing?
No
Set Dial mode to Pulse
Yes
Set Dial mode to Tone
Will this line be answered?
No
Set Answer mode to Auto
Will callers require a CoS password?
No
Do not select Answer with DISA
Yes
Select Answer with DISA
Yes
Set Answer mode to manual
Is there an external ringer that you want the line to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Wil this be a private line?
No
Do not select Auto privacy
Yes
Select Auto privacy
Is this line part of a private netwrok or will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedules for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Does this line connect to the remote voicemail system?
No
Line configuration complete
Yes
Select a configured voicemail center
Line configuration complete

Configuring digital ground start line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

To configure digital Ground Start line features

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Line: Unique number.
- Trunk type: Ground Start.
- Name: Identify the line or line function.
- Control set: Identify a DN if you are using this line with scheduling.
- Line type: Define how the line will be used. If you are using routing, ensure it is put into line pool (A to O). If you are using line pools, you must also configure target lines. ("Configuring lines: Target lines" on page 177)
- Prime set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.
- Distinct ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).
- Restrictions tab: Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

2 Configure the trunk/line data (Properties tab):

- Auto privacy: If you activate this feature, the line is available only to the telephone that answers the call.
- Dial mode: The line service will dictate whether this needs to be set to Pulse or Tone (DTMF) dialing.
- Answer mode/Answer with DISA: If this line is used for remote call-ins, determine how you want the line to answer (automatically, or requiring more user input). If the answer mode is set to Automatic, decide whether the caller will be immediately connected to the system or whether a stuttered dial tone will require the caller to enter a CoS password.
- Aux. ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- Redirect to: If you want to automatically direct calls out of the system to a specific telephone, such as a headoffice answer attendant, enter that remote number here. Ensure that you include the proper routing information.
- Voice Message Center: If the system is using a remote voice mail, pick the center configured with the contact number.

3 Set the restriction and remote package scheduling (Restrictions tab):

- Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)

- Remote Packages: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

4 Assign the lines to DNs (Assigned DNs tab)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs here. The DN record also can be used to assign lines and line pools for these lines.

- Appearance Type: Choose Appr only or Appr&Ring if the telephone has an available button, otherwise choose Ring only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

5 Suggested next steps:

- Dialing plan sections
  "Dialing plan: System settings" on page 303
  "Dialing plan: Public network" on page 311
  "Dialing plan: Routing and destination codes" on page 289
- Networking sections
  "Public networking: Setting up basic systems" on page 323
  "Public networking: Tandem calls from private node" on page 327
  "Private networking: Using shared line pools" on page 347
  "Private networking: Using destination codes" on page 373

Chapter 16
Configuring lines: T1-DID

DID (Direct Inward Dial) lines are lines on a digital trunk module on a T1. Inbound DID lines are mapped through target lines.

The following paths indicate where to access the DID lines in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Line**s

Task: Configure the properties for DID (Direct Inward Dial) lines

- "Configuring DID line features" on page 202

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

DTM module is installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 63 and Figure 64 provide an overview of the DID line features configuration process.

Figure 63 DID line feature configuration process — Part A

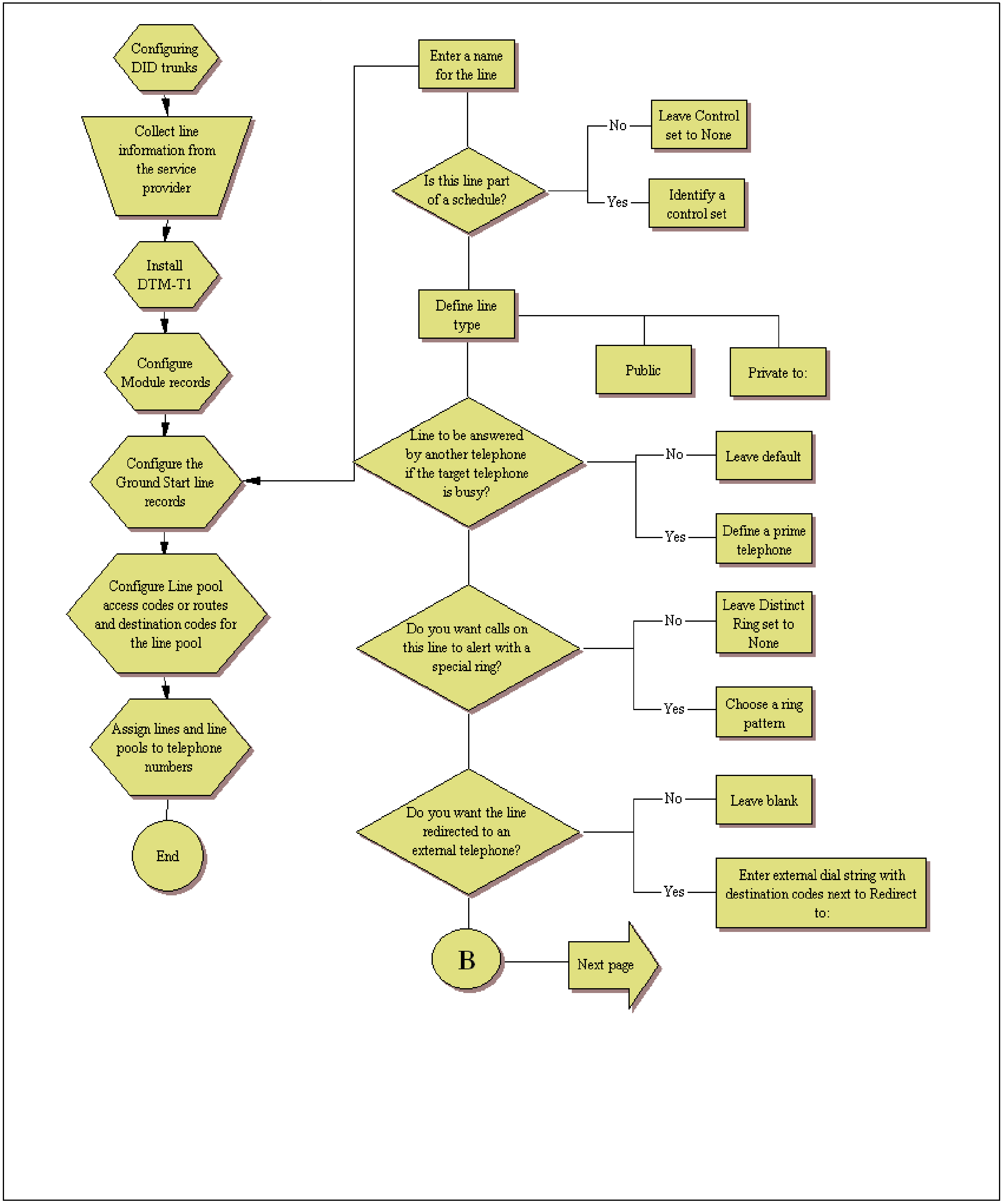
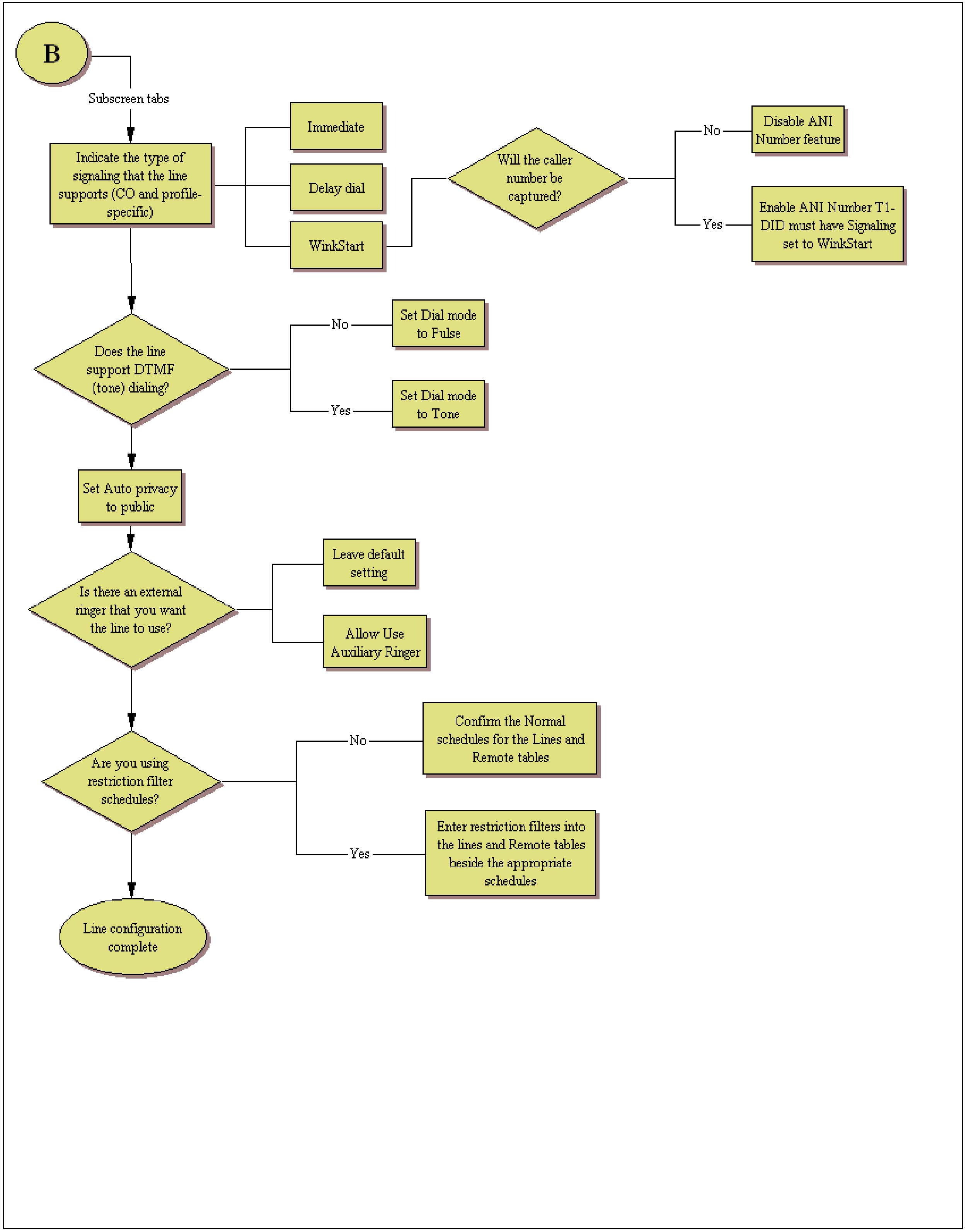

Configuring DID trunks
Collect line information from the service provider
Install DTM-T1
Configure Module records
Configure the Ground Start line records
Configure Line pool access codes or routes and destination codes for the line pool
Assign lines and line pools to telephone numbers
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set to None
Yes
Identify a control set
Define line type
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
Do you want the line redirected to an external telephone?
No
Leave blank
Yes
Enter external dial string with destination codes next to Redirect to:
B
Next page

Figure 64 DID line feature configuration process — Part B


B
Subscreen tabs
Indicate the type of signaling that the line supports (CO and profile-specific)
Immediate
Delay dial
WinkStart
Will the caller number be captured?
No
Disable ANI Number feature
Yes
Enable ANI Number T1-DID must have Signaling set to WinkStart
Does the line support DTMF (tone) dialing?
No
Set Dial mode to Pulse
Yes
Set Dial mode to Tone
Set Auto privacy to public
Is there an external ringer that you want the line to use?
Leave default setting
Allow Use Auxiliary Ringer
Are you using restriction filter schedules?
No
Confirm the Normal schedules for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Line configuration complete

Configuring DID line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

To configure DID line features

1. Confirm or change the settings on the Trunk/Line Data main panel:
    - Trunk Type: T-1 DID
    - Name: Identify the line or line function.
    - Control Set: Identify a DN if you are using this line with scheduling.
    - Line Type: Define as public if the line is shared, or as Private To (DN) if the line is assigned to a specific telephone.
    - Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
    - Pub. Received #: Not applicable.
    - Priv. Received #: Not applicable.
    - Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, or 4).
    - Use remote package: Not applicable.
2. Configure the trunk/line data (Properties tab):
    - Dial mode: The line service will dictate whether this needs to be set to Pulse or Tone (DTMF) dialing.
    - Signaling: Match this choice with the information supplied by the service provider.
3. Set the restriction and remote package scheduling (Restrictions tab):
    - Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
    - Remote Restrictions: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)
4. Assign the lines to DNs (Assigned DNs tab)(applicable to manual answer only)

    If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs, here. The DN record also can be used to assign lines and line pools for these lines.

    - Appearance type: Choose Appear or Appear and ring if the telephone has an available button, otherwise choose Ring Only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
    - VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system.

        Check with your system administrator for the system voice mail setup before changing this parameter.

5 Suggested next steps:

- Dialing plan sections
  “Dialing plan: System settings” on page 303
  “Dialing plan: Public network” on page 311
  “Dialing plan: Routing and destination codes” on page 289
- Networking sections
  “Public networking: Setting up basic systems” on page 323
  “Public networking: Tandem calls from private node” on page 327
  “Private networking: Using shared line pools” on page 347
  “Private networking: Using destination codes” on page 373

Chapter 17
Configuring lines: DASS2 lines

DASS2 lines are specific to the UK protocol.

The following paths indicate where to access the DASS2 lines in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset Interface: ****CONFIG > Lines**

Task: configure DPNSS lines connected to the system

- "Configuring DASS2 line features" on page 207
- Also refer to "Private networking: DPNSS network services (UK only)" on page 365

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

DTM module is installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 65 and Figure 66 provide an overview of the DASS2 line feature configuration.

Figure 65 DASS2 line feature configuration process — Part A

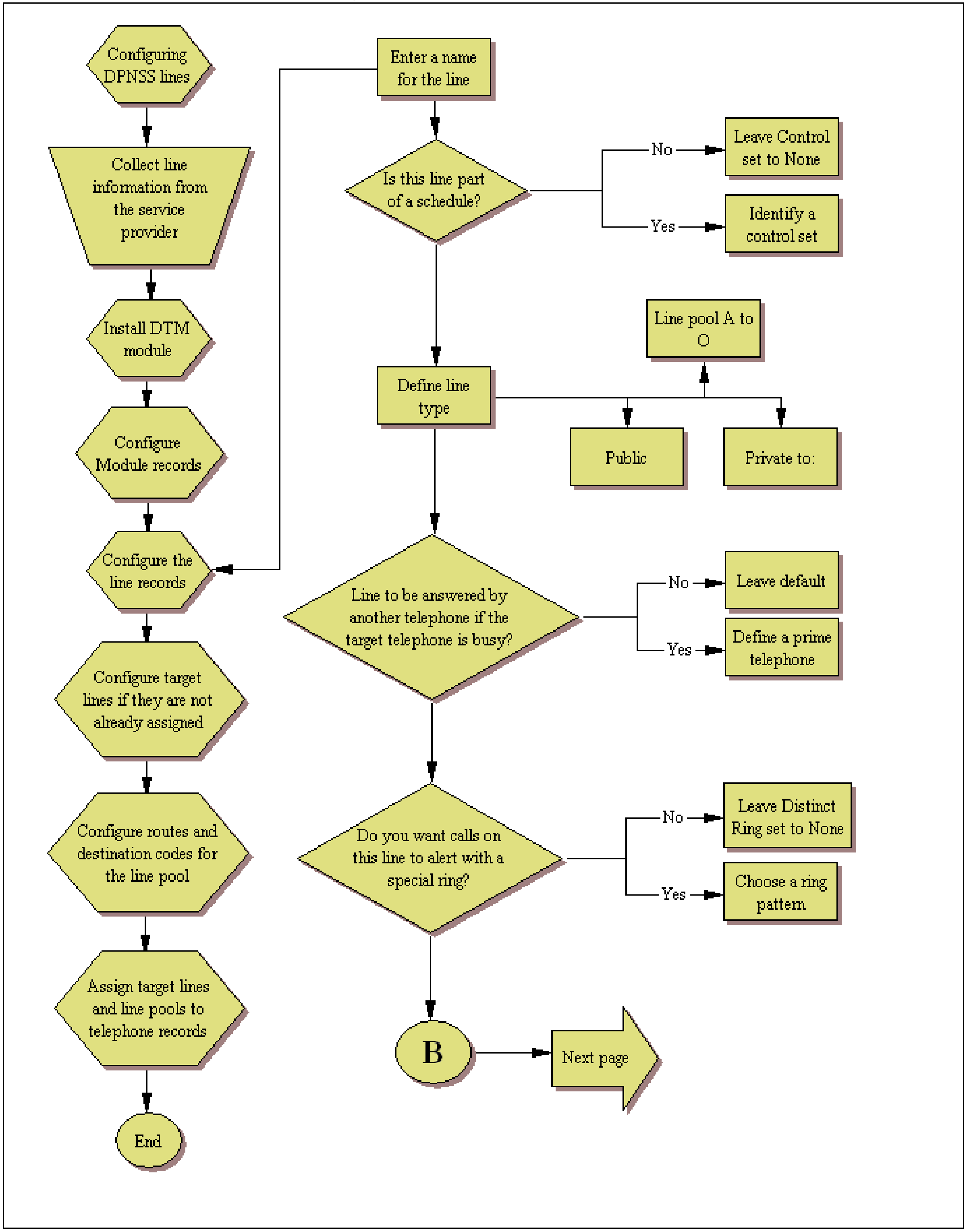
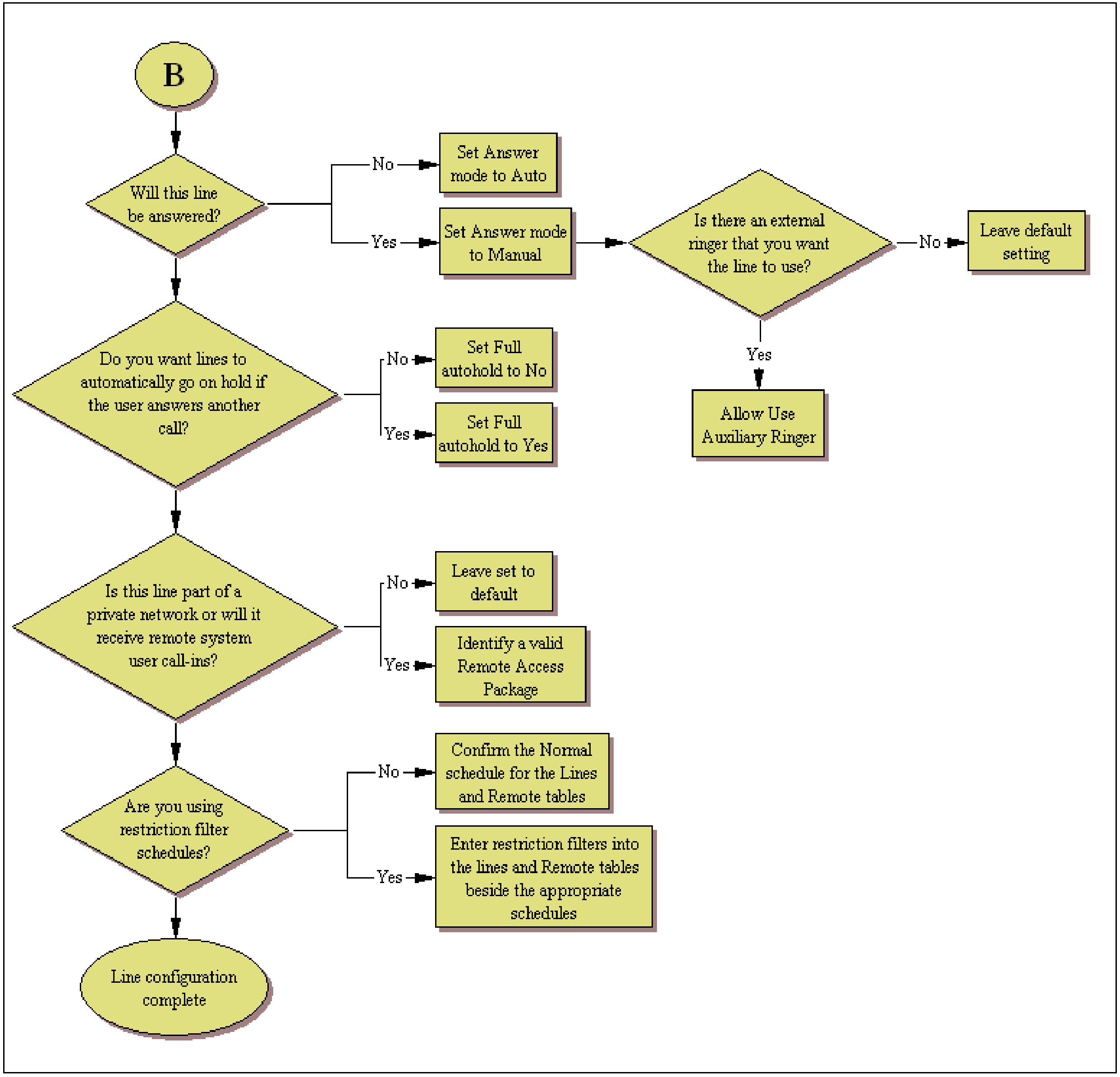

Configuring DPNSS lines
Collect line information from the service provider
Install DTM module
Configure Module records
Configure the line records
Configure target lines if they are not already assigned
Configure routes and destination codes for the line pool
Assign target lines and line pools to telephone records
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set to None
Yes
Identify a control set
Line pool A to O
Define line type
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
B
Next page

Figure 66 DASS2 line feature configuration process — Part B


B
Will this line be answered?
No
Set Answer mode to Auto
Yes
Set Answer mode to Manual
Is there an external ringer that you want the line to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Do you want lines to automatically go on hold if the user answers another call?
No
Set Full autohold to No
Yes
Set Full autohold to Yes
Is this line part of a private network or will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedule for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Line configuration complete

Configuring DASS2 line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

To configure DASS2 line features

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Trunk type: DASS2
- Name: Identify the line or line function.

- Control Set: Identify a DN if you are using this line with scheduling.
- Line type: Define how the line will be used. If you are using routing, ensure it is put into line pool (A to O).
- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.
- Distinct ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4 or None).
- Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

2 Configure the trunk/line data (Properties tab):

- Answer mode: If this line is used for remote call-ins, determine how you want the line to answer (automatically, or requiring more user input).
- Use auxiliary ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- Full autohold: This allows telephones to put a line on hold if the user picks up another line or starts to dial out on another line.
- Voice Message Center: If the system is using a remote voice mail, pick the center configured with the contact number.

3 Set the restriction and remote package scheduling (Restrictions tab):

- Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Packages: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

4 Assign the lines to DNs (Assigned DNs tab)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs, here. The DN record also can be used to assign lines and line pools for these lines.

- Appearance type: Choose Appr or Appr&ring if the telephone has an available button, otherwise choose Ring only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

5 Suggested next steps:

- Dialing plan sections

"Dialing plan: System settings" on page 303
"Dialing plan: Public network" on page 311
"Dialing plan: Private network settings" on page 317
"Dialing plan: Routing and destination codes" on page 289

- Networking sections
"Public networking: Tandem calls from private node" on page 327
"Private networking: Using destination codes" on page 373
"Private networking: DPNSS network services (UK only)" on page 365
"Private networking: MCDN over PRI and VoIP" on page 329

Chapter 18
Configuring lines: DPNSS lines

DPNSS lines are specific to the UK protocol.

The following paths indicate where to access the DPNSS lines in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines**
- Telset interface: ****CONFIG > Lines**

Task: configure DPNSS lines connected to the system

- "Configuring DPNSS line features" on page 213
- Also refer to "Private networking: DPNSS network services (UK only)" on page 365

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

DTM module is installed and configured. Refer to "Trunk Module Parameters" on page 122.	
Lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Process map

Figure 67 and Figure 68 provide an overview of the DPNSS line feature configuration process.

Figure 67 DPNSS line feature configuration process — Part A

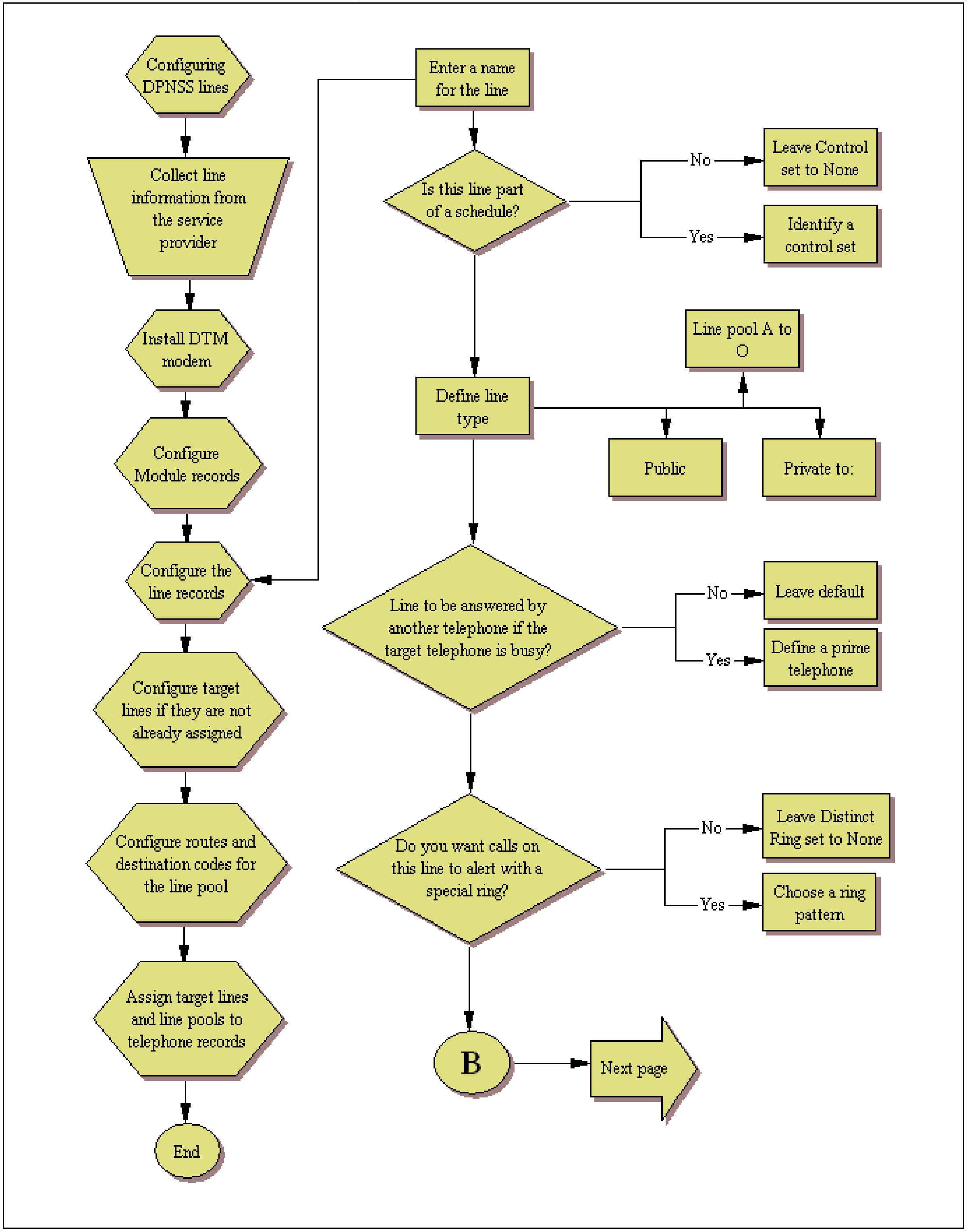
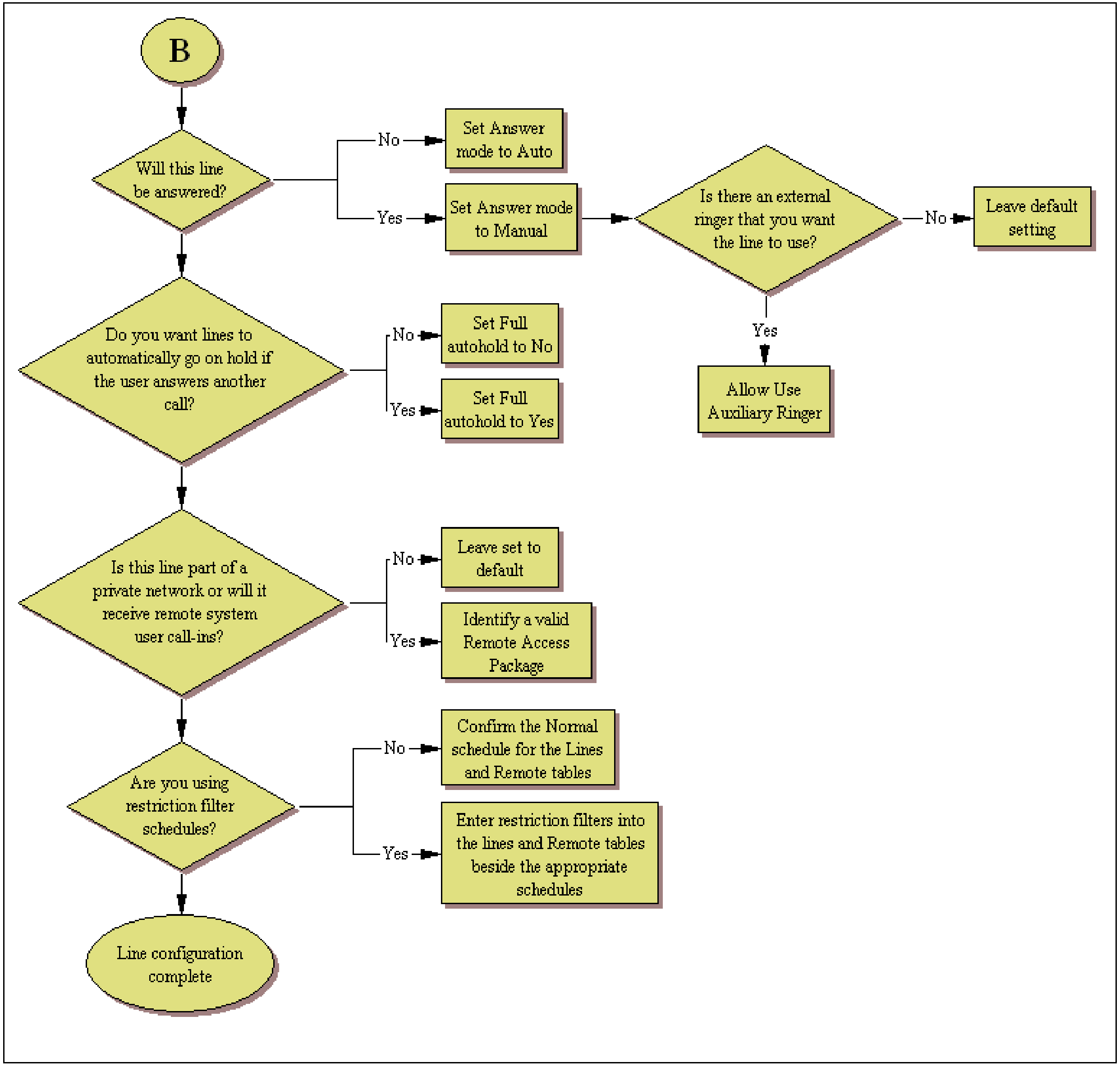

Configuring DPNSS lines
Collect line information from the service provider
Install DTM modem
Configure Module records
Configure the line records
Configure target lines if they are not already assigned
Configure routes and destination codes for the line pool
Assign target lines and line pools to telephone records
End
Enter a name for the line
Is this line part of a schedule?
No
Leave Control set to None
Yes
Identify a control set
Line pool A to O
Define line type
Public
Private to:
Line to be answered by another telephone if the target telephone is busy?
No
Leave default
Yes
Define a prime telephone
Do you want calls on this line to alert with a special ring?
No
Leave Distinct Ring set to None
Yes
Choose a ring pattern
B
Next page

Figure 68 DPNSS line feature configuration process — Part B


B
Will this line be answered?
No
Set Answer mode to Auto
Yes
Set Answer mode to Manual
Is there an external ringer that you want the line to use?
No
Leave default setting
Yes
Allow Use Auxiliary Ringer
Do you want lines to automatically go on hold if the user answers another call?
No
Set Full autohold to No
Yes
Set Full autohold to Yes
Is this line part of a private network or will it receive remote system user call-ins?
No
Leave set to default
Yes
Identify a valid Remote Access Package
Are you using restriction filter schedules?
No
Confirm the Normal schedule for the Lines and Remote tables
Yes
Enter restriction filters into the lines and Remote tables beside the appropriate schedules
Line configuration complete

Configuring DPNSS line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

To configure DPNSS line features

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Trunk type: DPNSS
- Control Set: Identify a DN if you are using this line with scheduling.

- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.

(Parameters tab)

- Name: Identify the line or line function.
- Line type: Define how the line will be used. If you are using routing, ensure it is put into line pool (A to O).
- Distinct ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4 or None).

(Preferences tab)

- Pub. Received #: Not applicable.
- Priv. Received #: Not applicable.
- Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

2 Configure the trunk/line data (Properties tab):

- Answer mode: If this line is used for remote call-ins, determine how you want the line to answer (automatically, or requiring more user input).
- Use auxiliary ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- Full autohold: This allows telephones to put a line on hold if the user picks up another line or starts to dial out on another line.
- Voice Message Center: If the system is using a remote voice mail, pick the center configured with the contact number.

3 Set the restriction and remote package scheduling (Restrictions tab):

- Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Packages: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

4 Assign the lines to DNs (Assigned DNs tab)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs, here. The DN record also can be used to assign lines and line pools for these lines.

- Appearance type: Choose Appr or Appr&ring if the telephone has an available button, otherwise choose Ring only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system. Check with your system administrator for the system voice mail setup before changing this parameter.

5 Suggested next steps:

- Dialing plan sections

"Dialing plan: System settings" on page 303
"Dialing plan: Public network" on page 311
"Dialing plan: Private network settings" on page 317
"Dialing plan: Routing and destination codes" on page 289

- Networking sections
  "Public networking: Tandem calls from private node" on page 327
  "Private networking: Using destination codes" on page 373
  "Private networking: DPNSS network services (UK only)" on page 365
  "Private networking: MCDN over PRI and VoIP" on page 329

Chapter 19
BRI ISDN: BRI T-loops

BRI modules support both trunk and station (telephone) services. The following describes the process for configuring trunk (T) loops.

Task: Configure BRI T-loops
"Configuring BRI T-loop parameters" on page 219

Prerequisites

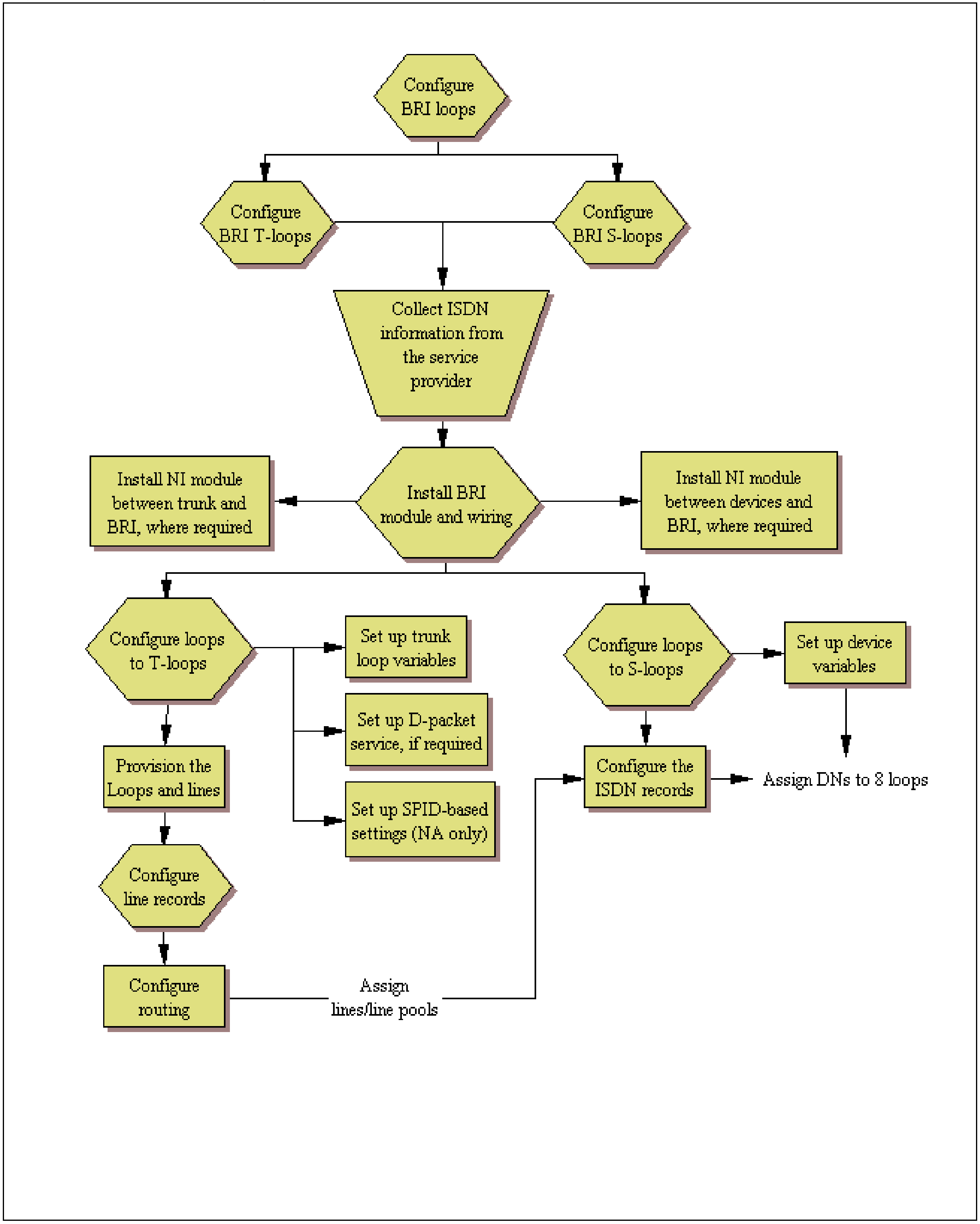
Complete the following prerequisites checklist before configuring the modules.

Ensure that system hardware is installed and operating correctly.	
Obtain all relevant central office/service provider information for the loops.	
BRI module is installed and operating (LEDs are correct).	

Process overview

Figure 69 shows the process for configuring BRI loops.

Figure 69 BRI loops configuration process


Configure BRI loops
Configure BRI T-loops
Configure BRI S-loops
Collect ISDN information from the service provider
Install NI module between trunk and BRI, where required
Install BRI module and wiring
Install NI module between devices and BRI, where required
Configure loops to T-loops
Set up trunk loop variables
Set up D-packet service, if required
Set up SPID-based settings (NA only)
Configure loops to S-loops
Set up device variables
Provision the Loops and lines
Configure the ISDN records
Assign DNs to 8 loops
Configure line records
Configure routing
Assign lines/line pools

Configuring BRI T-loop parameters

To configure BRI T-loop parameters

1. Identify the loop as a T-loop (refer to "Configure loop type and general parameters" on page 227).
   - Protocol (ETSI and ETSI-QSIG loops, only)
   - ONN block state
   - Overlap receiving
   - Overlap length
   - Send name display (ETSI-QSIG only)
2. Enter the details for the loop (refer to "T-loop SPIDS and network DNs" on page 229).

   North American systems only:
   - SPID
   - B-channel
   - Network DN
   - Call type

   ETSI and ETSI-QSIG T-loops (UK profile)
   - Clock source
3. If applicable, configure D-packet service for the loop (refer to "T-loops D-packet service" on page 231).
4. Provision the loop and the loop lines (refer to "Provisioning module lines/loops" on page 130).
5. Program the BRI lines (refer to "Configuring BRI lines" on page 219). If the lines are set to auto-answer, put the lines into line pools (A to O) and configure target lines.
6. Assign the lines/line pools and target lines to the telephones. Refer to Line Access - Line Assignment tab and Line Access - Line Pool Access tab in the *BCM 4.0 Device Configuration Guide* (N0060600).

Configuring BRI lines

There are two lines for every ISDN BRI loop that is designated as a T-loop. Unlike PRI lines, these lines can be set to either manual or automatic answer when using for remote call-ins.

The following paths indicate where to access the line configuration menu through Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Lines > Active Physical Lines, Inactive Lines, All Lines**
- Telset interface: ****CONFIG > Lines**

Prerequisites:

BRI module: Installed and configured. Refer to "Trunk Module Parameters" on page 122.	
BRI loops are configured as T loops. Refer to "Configuring BRI T-loop parameters" on page 219.	
BRI loop lines are provisioned. Refer to "Provisioning module lines/loops" on page 130.	

Configuring provisioned BRI line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to "Configuring lines" on page 147.

To configure provisioned BRI line features

1 Confirm or change the settings on the Trunk/Line Data main panel:

- Trunk Type: BRI-ST (determined by profile and type of BRI module)
- Name: Identify the line or line function.
- Control Set: Identify a DN if you are using this line with scheduling.
- Line Type: Define how the line will be used. If you are using routing, ensure it is put into line pool (A to O).
- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable.
- Priv. Received#: Not applicable.
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, or 4).
- Subpanel, under Restrictions tab: Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid package.

2 Configure the trunk/line data (Properties tab):

- Auto privacy: If you activate this feature, the line is available only to the telephone that answers the call.
- Answer mode/Answer with DISA: If this line is used for remote call-ins, determine how you want the line to answer (automatically, or requiring more user input). If the answer mode is set to Automatic, decide whether the caller will be immediately connected to the system or whether a stuttered dial tone will require the caller to enter a CoS password.
- Aux. ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.
- Full autohold: This allows telephones to put a line on hold if the user picks up another line or starts to dial out on another line.
- Voice Message Center: If the system is using a remote voice mail, pick the center configured with the contact number.

3 Set the restriction and remote package scheduling (Restrictions tab):

- Line restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Packages: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of (incoming calls from remote users or private networks)

4 Assign the lines to DNs (Assigned DNs tab)

If you have configured the DNs and know to which telephones the line needs to be assigned, you can enter those DNs, here. The DN record also can be used to assign lines and line pools for these lines.

- Appearance Type: Choose Appr only or Appr&Ring if the telephone has an available button, otherwise choose Ring only. Model 7000 and 7100 telephones have no programmable buttons, so this must be set to Ring only. (Model 7000 phones, supported in Europe only.)
- VMsg set: When activated, an indicator on the telephone appears when there is a message waiting from a remote voice mail system.

  Check with your system administrator for the system voice mail setup before changing this parameter.

Chapter 20

Programming BRI S-loops, lines and ISDN devices

BRI modules support both trunk and station (telephone) services. The following describes the process for configuring station/device (S) loops, which support devices that use an ISDN interface. You can assign a single device to a loop, or multiple devices connected through an NT-1 interface.

The following paths indicate where to configure loops through Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Loops**
- Telset interface: ****CONFIG > Hardware > Module > Loops**

Task: Configure BRI S-loops

- "Setting BRI properties for ISDN device connections" on page 223
- "DN records: ISDN devices" on page 224

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

Ensure that system hardware is installed and operating correctly.	
Obtain all relevant central office/service provider information for the loops.	
BRI module is installed and operating (LEDs are correct).	
Wiring is complete for ISDN device configuration.	

Setting BRI properties for ISDN device connections

BRI S-loops support devices that use an ISDN interface. Also refer to "ISDN overview" on page 701. You can assign a single device to a loop, or multiple devices connected through an NT-1 interface.

- You can assign a maximum of eight devices to a loop.
- Any device can only be configured to one loop.
- S-loops do not supply any voltage for ISDN devices requiring power, such as video cameras. Voltage for these devices must be supplied by an external source on the S-loop.

For detailed descriptions of the BRI module fields, refer to "BRI ISDN: BRI loop properties" on page 227.

To set BRI properties for ISDN device connections

1. On the top panel, identify the loop as an S-loop. Refer to "Configure loop type and general parameters" on page 227.
   - Sampling
2. On the bottom panel, identify which ISDN DNs to associate to the loop (see "S-loops assigned DNs" on page 232) (Default ISDN DNs: 365-396):
   - Assigned DNs
   - Loop DN (must be on the Assigned DN list). If you set this field to None, unanswered calls are dropped. If the field is left blank, Assigned DNs make and receive data calls.
3. Configure the ISDN DN records for the device(s) assigned to the loop. Refer to "Configuring an ISDN telephone DN record" on page 226.

DN records: ISDN devices

ISDN telephones and devices have a limited feature set. They do not have programmable buttons or user preferences, and do not support call forward features. However, you can assign Answer DNs and some capabilities features.

Task: Determine the programming for individual telephones and devices attached to BRI module S-loops.

- "Configuring an ISDN telephone DN record" on page 226
- "ISDN overview" on page 701

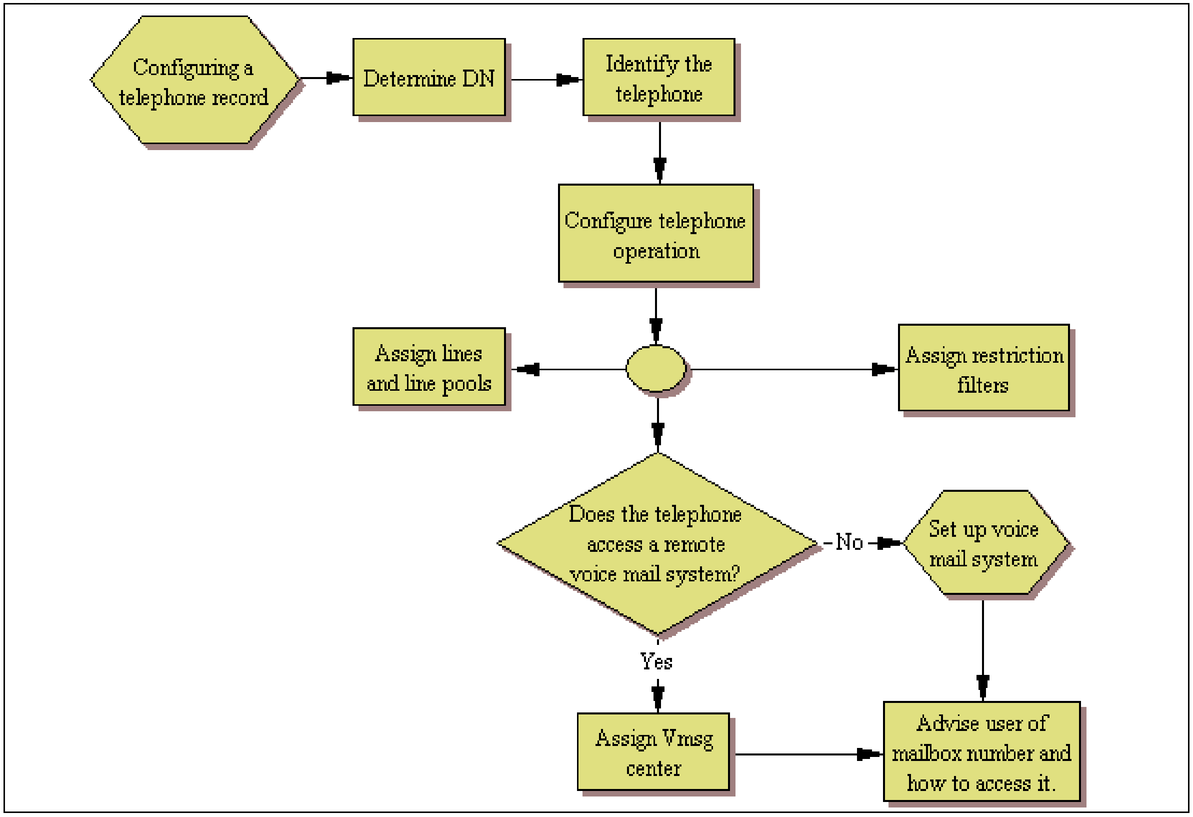
For a detailed description of DN record panels, and DN record procedures, refer to "DN records parameters" in the *BCM 4.0 Device Configuration Guide* (N0060600).

ISDN devices have a DN range that is unique to ISDN devices.

Process map

Figure 70 provides an overview of the ISDN DN record configuration process.

Figure 70 ISDN DN record overview


Configuring a telephone record
Determine DN
Identify the telephone
Configure telephone operation
Assign lines and line pools
Assign restriction filters
Does the telephone access a remote voice mail system?
No
Set up voice mail system
Yes
Assign Vmsg center
Advise user of mailbox number and how to access it.

Prerequisites

Complete the following prerequisites checklist before configuring the devices.

BRI module installation and configuration is complete. Refer to “Trunk Module Parameters” in the *BCM 4.0 Device Configuration Guide* (N0060600).	
BRI loops programming is complete. Refer to “Setting BRI properties for ISDN device connections” on page 223.	
Lines are provisioned and configured. Refer to “Provisioning module lines/loops” in the *BCM 4.0 Device Configuration Guide* (N0060600).	
Wiring and network connections for the devices are complete.	

Configuring an ISDN telephone DN record

On each panel on the DNs list, add or modify settings to customize the telephone operations. The following headings correspond to each panel. Refer to the **Programming notes** in each section for configurations that are unique or specific for ISDN telephones.

Table 47 ISDN device-specific DN record settings

Affected field	Setting	Panel name and link to common procedures
Name	Unique to each device or device loop	"System DNs - Line Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600)
Call forward	Not supported	
Line appearances	Ring only	"Line Assignment and Line Pools" in the *BCM 4.0 Device Configuration Guide* (N0060600)
Answer DNs	Ring only	
Intercom keys	two: not configurable	"Configuring Capabilities and Preferences" in the *BCM 4.0 Device Configuration Guide* (N0060600)
The following settings are the only capability settings that require specific configuration for ISDN devices.		
Page settings	Page only- select. Devices cannot be assigned to Page zones.	"Configuring telephone capabilities" in the *BCM 4.0 Device Configuration Guide* (N0060600)
OLI as called number	<check box>	If Enabled, the specified OLI for the telephone is used for CLID for calls.
All other settings are variable, based on your system requirements.		

Chapter 21
BRI ISDN: BRI loop properties

The Loops tables display settings for installed BRI modules.

The following paths indicate where to access the loops table for BRI modules in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Loops**
- Telset interface: ****CONFIG > Hardware > Module > Loops**

This panel contains the following tab:

- Loops - provides configuration for general loop settings.

Click one of the following links to access the type of information you want to view:

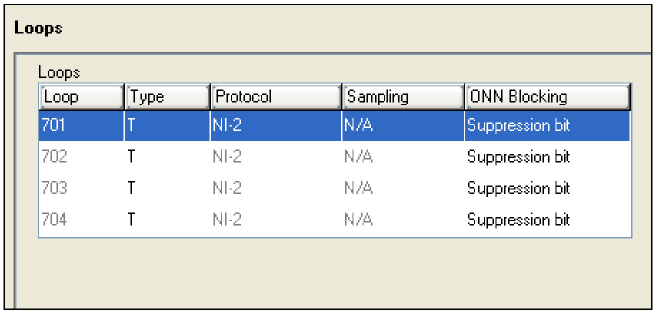
Panel tabs	Tasks
“Configure loop type and general parameters” on page 227	ONN blocking
“T-loop general settings” on page 229	“Provisioning module lines/loops” on page 130 “Configuring lines” on page 147 “Configuring lines: T1-Loop start” on page 187
“T-loop SPIDS and network DNs” on page 229	“BRI ISDN: BRI T-loops” on page 217
“S-loops assigned DNs” on page 232	“Programming BRI S-loops, lines and ISDN devices” on page 223 “DN records parameters” in the *BCM 4.0 Device Configuration Guide* (N0060600)
“T-loops D-packet service” on page 231	
Click the navigation tree heading to access general information about user management.	

You can define BRI loops as either T-loops, for connecting to ISDN trunks, or S-loops, for connecting to internal ISDN equipment. Both types of loops are displayed in the top frame in the Loop Parameters panel. In the bottom frame, the settings displayed are specific to each type of loop.

Configure loop type and general parameters

The Loops table displays the BRI loops for an installed module and the settings that are common to both T-loops and S-loops. Figure 71 illustrates the Loops table.

Figure 71 Loops table



Loops

Loops

Loop	Type	Protocol	Sampling	ONN Blocking
701	T	NI-2	N/A	Suppression bit
702	T	NI-2	N/A	Suppression bit
703	T	NI-2	N/A	Suppression bit
704	T	NI-2	N/A	Suppression bit

Table 48 describes the fields found on the Loop main panel.

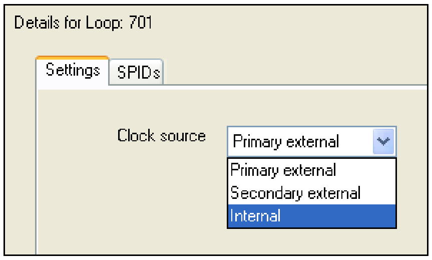
Table 48 Loops main panel

Attribute	**Value**	**Description**
Loop	<X01-X04>	Each BRI module supports four loops (eight lines for T-loop programming).
Type	T S	This setting defines whether the loop supports trunks (T-loop) or device connections (S-loop). **Note**: This variable may be different for different market profiles.
Protocol	Euro QSIG NI-2	Select the appropriate ISDN protocol. The values displayed depend on both the market profile and software keycodes. Euro - ETSI ISDN standard QSIG - also an ETSI standard. Only appears if the ETSI QSIG keycode is loaded. NI-2
Sampling (S-loops only)	Adaptive Fixed N/A	Select a sampling rate for the S-loop. Fixed: two or more S-interface devices use the loop, and the length of the loop is less than 200 m (650 ft.). Adaptive: two or more S-interface devices use the loop, and the length of the loop is greater than 200 m (650 ft.). If one device is using the loop, the length of the loop can be a maximum of 1000 m (3230 ft)
ONN blocking	Suppression bit Service code N/A	Set the Outgoing Name and Number (ONN) Blocking. When you activate ONN, a user can press **FEATURE 819** to block the outgoing name and number on a per call basis. **Programming note**: Ensure that all telephones that have this feature available are assigned valid OLI numbers. Refer to "Programming outgoing number display (OLI)" on page 239.
ONN blocking		**Suppression bit:** the system flags the call to the Central Office (CO) so that the name and number is not sent to the person you call. **Service code:** VSC digits are dialed out before the called number to activate ONN at the central office. These codes are supplied by your service provider for the lines. Refer to "ONN Blocking codes (North American systems)" in the *BCM 4.0 Device Configuration Guide* (N0060600). PRI lines have only one code, so do not require specific configuration.

T-loop general settings

The Settings tab allows you to define loop characteristics. Note that not all of these settings are required in all BRI markets. Figure 72 illustrates the Settings tab.

Figure 72 Settings subpanel (T loops)



Details for Loop: 701

Clock source: Primary external

Table 49 describes the fields on this panel.

Table 49 Details for Loop

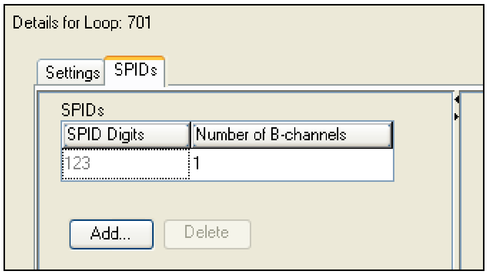
Attribute	**Value**	**Description**
Clock source	Primary External Secondary External Internal	Primary External - uses clock from PSTN Secondary External - used if system has more than one Loop Internal - uses clock on BCM
Overlap: receiving	<check box>	Supports target lines in markets which use Overlap receiving signaling on the BRI trunks. Overlap receiving must be configured for each BRI loop.
Overlap: length	0-10	Set the local number length for loops to interfaces that receive overlap rather than enbloc digits. This number is the total length of the called party number received. This number is used to calculate the number of leading digits that need to be removed by the system.
	Note: This parameter appears only when Overlap receiving is enabled. Example: Public received number = 4502303 Target line received numbers = 303 Local number length = 7 Public received number length = 3 Thus the first four digits are deleted by the system.	
Send Name Display (ETSI QSIG only)	<check box>	If the switch allows outgoing name display, select the check box.

T-loop SPIDS and network DNs

The T-loop SPIDS and network DNs settings are available only for systems running a North American profile. SPID numbers are supplied by the ISDN service provider. Also refer to “ISDN overview” on page 701.

Figure 73 illustrates the SPIDs tab.

Figure 73 SPIDs and network DNs (T-loops, North America only)



Details for Loop: 701

SPIDs

SPID Digits	Number of B-channels
123	1

Table 50 defines the fields on the SPIDs tab and indicates the lines.

Table 50 Loop settings (Sheet 1 of 2)

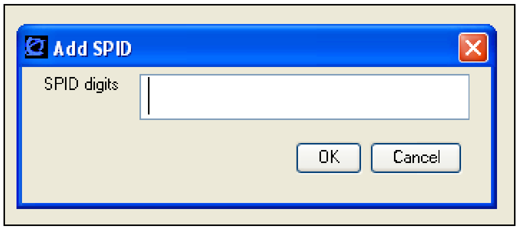
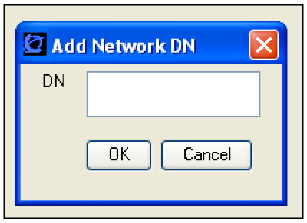
Attribute	Value	Description
SPIDS table		
SPID Digits	<digits>	Supplied by your service provider. System running with North American country profiles support additional BRI services offered by ISDN service providers and defined by network service profile identifiers (SPID). The SPID allows you to enter a network connection that provides a path for voice or data services.
Number of B-channels	1, 2	North American BRI loops can support two B-channels. The SPID may be the same or different for the channels.
Actions		
Add (SPID digits)	1. Click **Add**.  Add SPID SPID digits 2. Enter the SPID digits supplied by your ISDN service provider. 3. Click **OK**. 4. On the table, click the Number of B-channels field beside the number you entered. 5. Choose the number of B-channels allowed for this SPID.	
Delete	1. Select the SPID that you want to delete. 2. Click **Delete**. 3. Click **OK**.	
Network DNs table		
DN	<system DN>	This ISDN DN acts as the contact point for the loop to the system.
Call Type	Voice Data Both	Defines the type of calls supported on the loop.

Table 50 Loop settings (Sheet 2 of 2)

Attribute	Value	Description
Actions		
Add	1. Select the Network DN that you want to delete. 2. Under the Details for SPID table, click **Add**.  3. Enter a network DN. 4. Click **OK**. 5. On the table, click in the Call Type field beside the DN you entered. 6. Choose the call type for the DN.	
Delete	1. Select the Network DN that you want to delete. 2. Click **Delete**. 3. Click **OK**.	

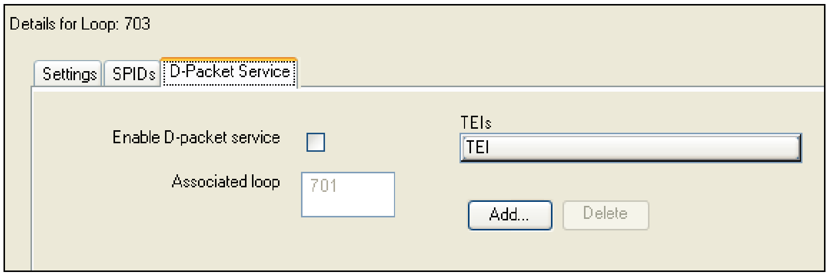
T-loops D-packet service

> **Note:** D-Packet service is only available if your service provider provides this Capability.

Use the D-Packet service tab to configure D-Packet Service to T-loops. You must have both T-loops and S-loops configured on the same module to allow this feature.

Figure 74 illustrates the D-Packet Service panel.

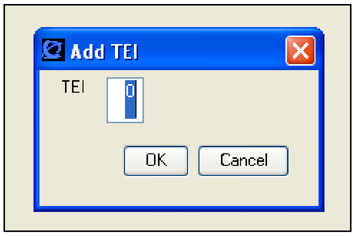
Figure 74 D-Packet Service (T-loops) tab



Details for Loop: 703

Table 51 describes each section on the D-Packet Service panel.

Table 51 D-packet settings

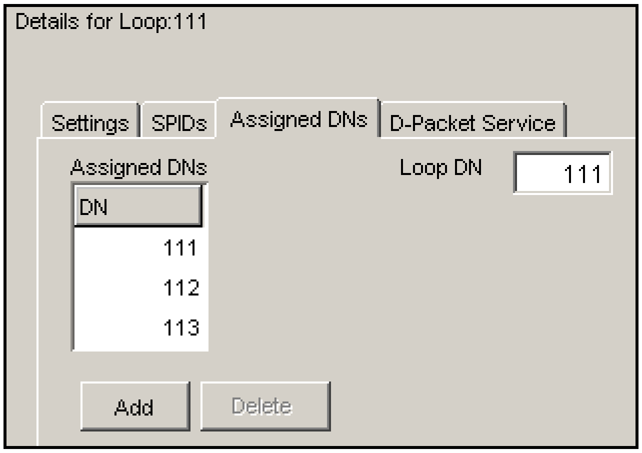
Attribute	Value	Description
Associated loop	X01-X04	S-loop: This is the loop on the BRI module where the device is connected.
Enabled D-packet Service	<check box>	Enable this service, only if you are installing devices that require this type of service.
TEI	<digits>	These entries identify up to eight terminal identifiers for the devices assigned to the S-loops. Your BRI service provider supplies these numbers, if they are required.
Actions		
Add	1. In the top frame, click the loop where you want to define D-Packet Service. 2. In the bottom frame, Ensure Enable D-packet service check box is selected. 3. In the Associated loop field, enter a defined S-loop. 4. Under the TEIs table, click **Add**.  Add TEI 5. Enter a TEI. 6. Click **OK**. 7. Repeat for all the TEIs you want to assign.	
Delete	1. In the top frame, click the loop where you want to delete TEI assignments. 2. In the bottom frame, click the TEI you want to delete. 3. Click **Delete**. 4. Click **OK**.	

S-loops assigned DNs

The Details for Loop panel for S-loops allows you to view which device records are assigned to a loop, and to add or delete a record from the loop.

Figure 75 illustrates the Details for Loop panel.

Figure 75 Assigned DNs (S-loops)



Details for Loop:111

Assigned DNs

DN
111
112
113

Loop DN 111

Table 52 defines the fields on the Details for Loop panel.

Table 52 Loop settings

Attribute	Value	Description
Loop DN	<system DN>	Control DN for the loop. This DN must be on the Assigned DNs list.
Assigned DNs table		
DN	<system DN>	ISDN assigned to the loop (up to eight devices)
Actions		
Add	1. In the top frame, click the loop where you want to add DN records. 2. In the bottom frame, click **Add**. 3. Enter the DN record number. 4. Click **OK**. 5. Repeat for all the DN records you want to assign.	
Delete	1. In the top frame, click the loop where you want to delete DN record assignments. 2. In the bottom frame, click the DN record you want to delete. 3. Click **Delete**. 4. Click **OK**.	

Chapter 22
Configuring CLID on your system

The following describes the various areas in the system that need configuration to allow incoming or outgoing Calling Line Identification Display (CLID) information to display (incoming calls) or transmit over the trunks (outgoing calls).

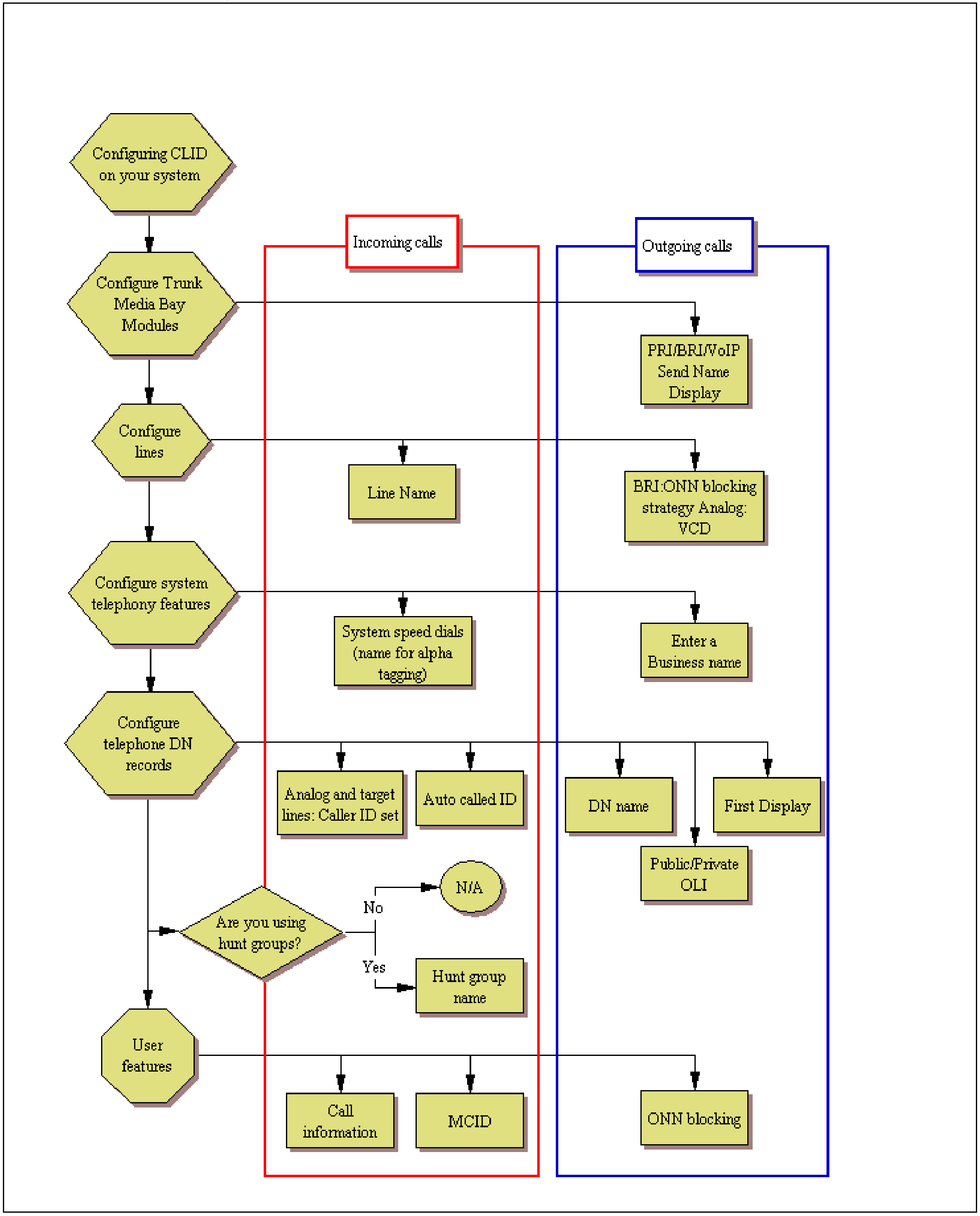
The following describes programming and setting up this feature.

Tasks:
Set up incoming display: "Programming incoming CLID" on page 237
Set up outgoing display: "Programming outgoing CLID" on page 238
Set up the method for blocking outgoing set identification: ONN Blocking (North American systems)" in the *BCM 4.0 Device Configuration Guide* (N0060600)

Process map

Figure 76 provides a quick view of the areas of the system that require programming to provide incoming and outgoing CLID services.

Figure 76 CLID configuration process


Configuring CLID on your system
Incoming calls
Outgoing calls
Configure Trunk Media Bay Modules
PRI/BRI/VoIP Send Name Display
Configure lines
Line Name
BRI:ONN blocking strategy Analog: VCD
Configure system telephony features
System speed dials (name for alpha tagging)
Enter a Business name
Configure telephone DN records
Analog and target lines: Caller ID set
Auto called ID
DN name
First Display
Public/Private OLI
N/A
No
Are you using hunt groups?
Yes
Hunt group name
User features
Call information
MCID
ONN blocking

Programming incoming CLID

Telephones can receive Name, Number, and Line display for incoming calls over trunks that support CLID or between telephones within the system. The following describes the different areas where these capabilities are configured.

➡ **Note:** If no configuration is done, CLID will show up after answering a call unless **Feature 811** is used. To make CLID appear before answer, you must set the Caller ID set on the set programming.

Digital, analog, and VoIP lines support CLID for incoming calls, and there is no special programming to allow the feature on these lines for BCM digital or IP phones.

Allowing CLID for telephones (incoming)

Target lines and analog CLID trunks connected to a GATM:

1. Under **Configuration > Telephony > Sets > Active Sets > Line Access**, select the DN record for a telephone assigned with analog lines that support CLID.
2. On the Line Assignment table, select a line that supports CLID.
3. Select the check box beside the Caller ID set field of the highlighted row.
4. Repeat the procedure for each line assigned to the telephone.
5. Repeat above steps for telephones assigned with these lines.

➡ **Note:** Only 30 telephones can be assigned CLID for a line.

Using alpha tagging for name display (incoming)

To set up alpha tagging on your system

1. To determine the name to display, you add a system speed dial for the number, entering a display name. Refer to “Configuring system speed dial numbers” in the *BCM 4.0 Device Configuration Guide* (N0060600).

➡ **Note:** You can increase the default number of system speed dials from 70 to 255 if you want to provide an extensive CLID list.

2. To determine how many digits of the dialed number and the system speed dial must match before a name is displayed, you set the **Clid match length** setting to the required number (1 to 8).

3 In order for the telephone to display the name, it must have **Caller ID** set for the line assigned to the telephone. Refer to "Line Access - Line Assignment tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

4 As well, **First display** must be set to **Name.** Refer to "Capabilities and Preferences main tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Programming line name display (incoming)

Answered calls can display the name, incoming number, and line name/number for calls coming in over lines that allow full CLID.

Lines are named by their number as a default. However, you can provide a more descriptive identifier. The Name field is located on the main table under **Telephony > Lines** ("Trunk/Line Data, main panel" in the *BCM 4.0 Device Configuration Guide* (N0060600)).

On the Hunt group record (**Telephony > Hunt Groups > Hunt Groups table**), you can change the Hunt group Name field from the Hunt group DN to a more logical label for the group. Note that only eight characters display. Refer to "Hunt Groups system setup" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Programming outgoing CLID

Telephones can transmit a business name, telephone name and number (outgoing line identifier) for outgoing calls over trunks to switches that support outgoing name and number (ONN) display, or between telephones within the system. This section describes the different areas where these capabilities are configured.

Programming Business name display (outgoing)

Nortel recommends that you use a blank space for the last character of the Business name to act as a separator between the Business name and telephone name.

Note that if you leave this field blank, no name appears.

To program the Business Name, select **Configuration > Telephony > Global Settings > General > Global Telephony Settings panel** > top panel.

To program the Business Name

1 Click the field beside Business Name.

2 Type a maximum of eight characters for a name.

Leave a blank space for the last character of the Business name to act as a separator between the Business name and telephone name.

3 Other areas that you must program include:

- The **OLI number**. Refer to “Line Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).
- The **Auto Called ID** must be selected. Refer to “Capabilities and Preferences - Capabilities tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Internal name and extension display

If you want to be able to see the CLID of internal telephones you call, ensure that Auto caller ID is enabled; select **Configuration > Telephony > System DNs > Capabilities and Preferences**. Refer to “Capabilities and Preferences main tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Programming name display (outgoing)

You can program name display for individual telephones.

On the DN record, you can change the Name field from the DN to a more logical label (Telephony > System DNs > System DNs table > any tab). Note that only eight characters appear. Refer to “Main panel tabs: common fields” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Programming outgoing number display (OLI)

You can determine what number displays at the other end of an outgoing call, if the outgoing line allows name display and the receiving telephone has number display active.

Note: OLI is not supported on analog trunks.

The Outgoing Line Identification (OLI) can be set for each telephone for both private and public network calls.

The Private OLI is used for CLID over private networks. It is usually set to the DN number as a default, although this does not always occur if there have been DN length changes. (Select **Configuration > Telephony > Sets > Active Sets > Line Access table**). Refer to “Line Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600). If the system is running with a UDP dialing plan, you might want to add the LOC to the DN. Refer to “Outgoing private calls routing” on page 322.

The Public OLI is used for CLID over public networks and for tandemed calls over private networks that terminate on the public network. The number of digits for this field is determined by your local service provider. (Select **Configuration > Telephony > Sets > Active Sets > Line Access table**). Refer to “Line Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Blocking outgoing name display at the trunks

To block outgoing name display at the media bay module level, you can configure module records to disable the Send Name display check box. To access the check box, select **Configuration > Resources > Telephony Resources > Trunk Module Parameters** (not available for all trunk types). Refer to “Trunk Module parameters” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Blocking outgoing name display at the telephone

ONN is also enabled and disabled from a telephone, on a per-call basis, using **FEATURE 819**.

To allow **FEATURE 819** to work correctly, you may need to specify an ONN blocking service code.

The BCM alerts the CO by two methods. The method used depends on the type of trunk involved in placing the outgoing call. This information is supplied by your service provider.

- Analog trunks use a dialing digit sequence called a Vertical Service Code (VSC). The VSC differs from region to region and must be programmed. Analog trunks with both tone and pulse dialing trunks can have separate VSCs.
- PRI trunks have only one VSC. No specific system programming is required.

  ETSI note: ETSI lines may use the Calling Line Information Restriction (CLIR) supplementary service to provide this feature.
  ETSI PRI lines do not use a VSC. The line always uses Suppression bit to invoke the CLIR supplementary service.

- BRI trunks can be set to either:
  - — provide ONN using a suppression bit, which provides a notice from the system to the central office to withhold CLI.
  - — provide ONN using a VCS, which is dialed out in front of the dialed digits (optional on ETSI trunks).

  BRI trunk ONN settings are located under the loops settings. Refer to “BRI ISDN: BRI T-loops” in the *BCM 4.0 Device Configuration Guide* (N0060600).

Programming note: Ensure that users who have access to this feature have telephones with valid OLI numbers.

Chapter 23
CLID: Name display

BCM displays the name of the calling party at the answering telephone when this information is available on Private or Public PRI trunks, BRI trunks, VoIP trunks, and analog trunks that support Calling Line Identification (CLID). The displayed name can include the Receiving Calling Name, Receiving Redirected Name, and/or Receiving Connected Name. Refer to "Receiving and sending calling party name" on page 241.

If only a number is available for CLI on an incoming call, you can program a system speed dial in such a way that a name displays when that number calls in. Refer to "Alpha tagging for name display" on page 242.

Name and number information is also transmitted with outgoing calls. This can be blocked by the user (**FEATURE 819**) on a per-call basis. As well, you can block this information on a per-trunk basis. This is important if the connecting system cannot process name and number information. Some service providers also may have different codes that need to be mapped so that the blocking feature works.

Table 53 provides a list of the name/number display features and the list of ISDN interfaces that support each feature.

Table 53 Call features/interface list

	Interface					
Feature	**NI PRI**	**DMS Custom PRI**	**SL-1 (MCDN)**	**NI-BRI**	**ETSI Euro (PRI/BRI)**	**ETSI QSIG**
Receiving Calling Name	Supported	Supported	Supported	Supported		Supported
Receiving Redirected Name	Supported		Supported	Supported		
Receiving Connected Name		Supported	Supported			Supported
Sending Calling Party Name	Supported	Supported	Supported			Supported
Sending Connected Name		Supported	Supported			Supported

Note: Name Display is an optional feature that is available based on the interface to which you subscribe.

Note: MCDN networks fully support name display features within the private network environment.

Receiving and sending calling party name

Network Name Display allows the name of an incoming PRI/BRI, analog with CLID, or VoIP with MCDN call to appear on the BCM telephone receiving the call.

Calling Party Name with status of Private can appear on the Called Party telephone as Private name. If the incoming Calling Name is defined by the CO as a private name, then Private name appears on the answering telephone. If the Calling Party Name is unavailable it can appear on the Called Party telephone as Unknown name.

If the call is answered by a Hunt group, the hunt group name appears instead of the telephone name in forming the connected name.

The Connected Name is a transient display that appears for approximately three seconds. The Connected Name is sent only if the OLI is programmed. You can program both a public and private OLI. The system uses the one appropriate to the type of call.

Network name display interactions

Calling and Connected Name information (if available) passes between trunks with Selective Line Redirection (SLR). Only Calling Name information passes between trunks in cases where Direct System Inward Access (DISA) results in tandeming of trunks.

Outgoing name display

You can set up the trunks to disallow name display to be sent out on PRI, BRI, and VoIP trunks. Use this for trunks where the connecting switch does not support outgoing line display. Default is enabled.

Business name display

Nortel recommends that you use a blank space for the last character of the Business name to act as a separator between the Business name and telephone name. A maximum of eight characters is supported.

Alpha tagging for name display

You can configure your system to display a caller name for incoming calls that provide number-only CLID, such as if the name service is not subscribed to or available in your area.

➡ **Note:** Lines that provide name and number CLID, such as PRI lines, use that name for display, rather than the alpha tagging feature.

Limitations:

- Due to system resource limitations, only 30 telephones can be assigned to provide alpha tagging CLID per line.
- If the incoming number only partially matches the CLID match length, no name displays.
- If the number matches more than one speed dial, and the matches have different names, the telephone displays the name of the first match.
- ISDN devices do not support the alpha tagging feature.

Name display

You can assign names to identify your company, external lines, target lines, and your colleagues' telephones. During a call, the name (if programmed) appears on the telephone display instead of on the external line number or internal telephone number of the caller.

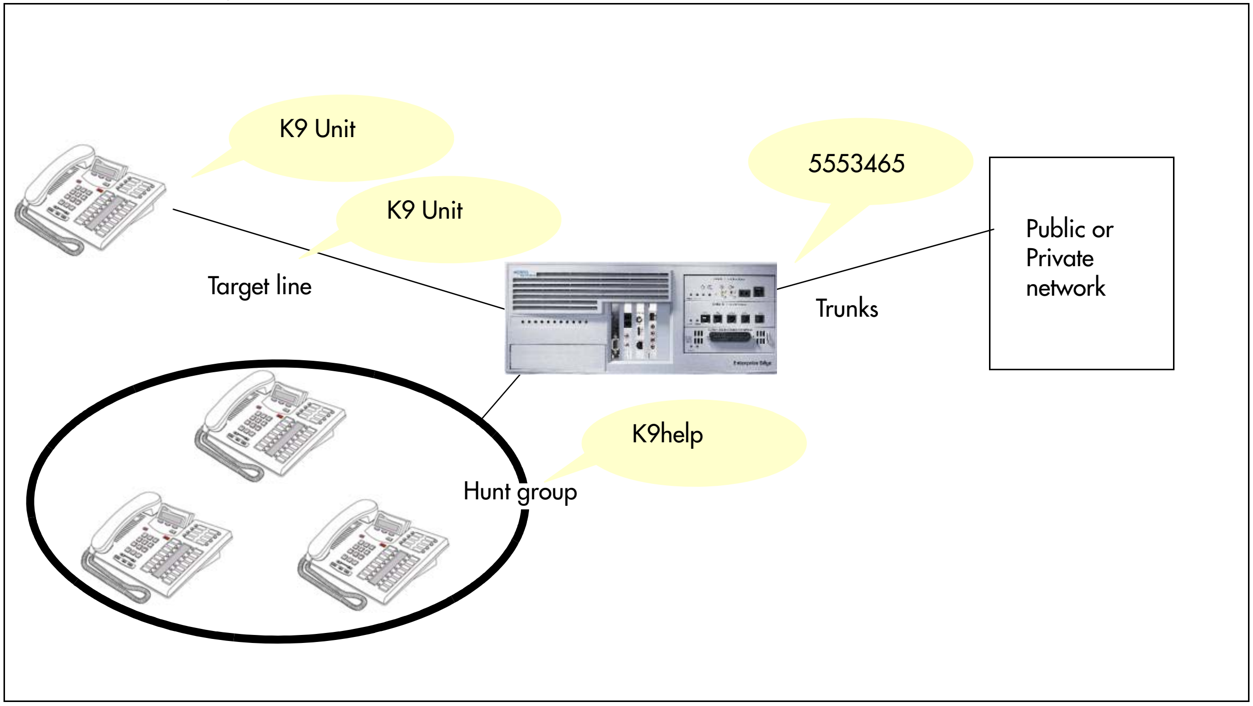
Names can contain both letters and numbers, but cannot be longer than seven characters. You cannot use the number (#) and star (*) symbols.

Note: You can give the same name to a telephone and a line in your system. Use initials, abbreviations, or even nicknames to give each telephone a unique name to avoid confusion.

You can also determine if the calling line ID (CLID) is received by a telephone, or if the CLID information from a system telephone gets sent out over the network. Refer to "Incoming and outgoing call display" on page 244.

Figure 77 illustrates an example of naming system components.

Figure 77 Naming components in the system



Incoming and outgoing call display

If you subscribe to Call Display services from your local telephone company, one line of information about an external caller appears on the display after you answer a call. If you answer before the Call Display information appears on your display, press **FEATURE 811** to view the line number or line name. When you transfer an external call to another telephone in your system, the same information appears on the recipient telephone display.

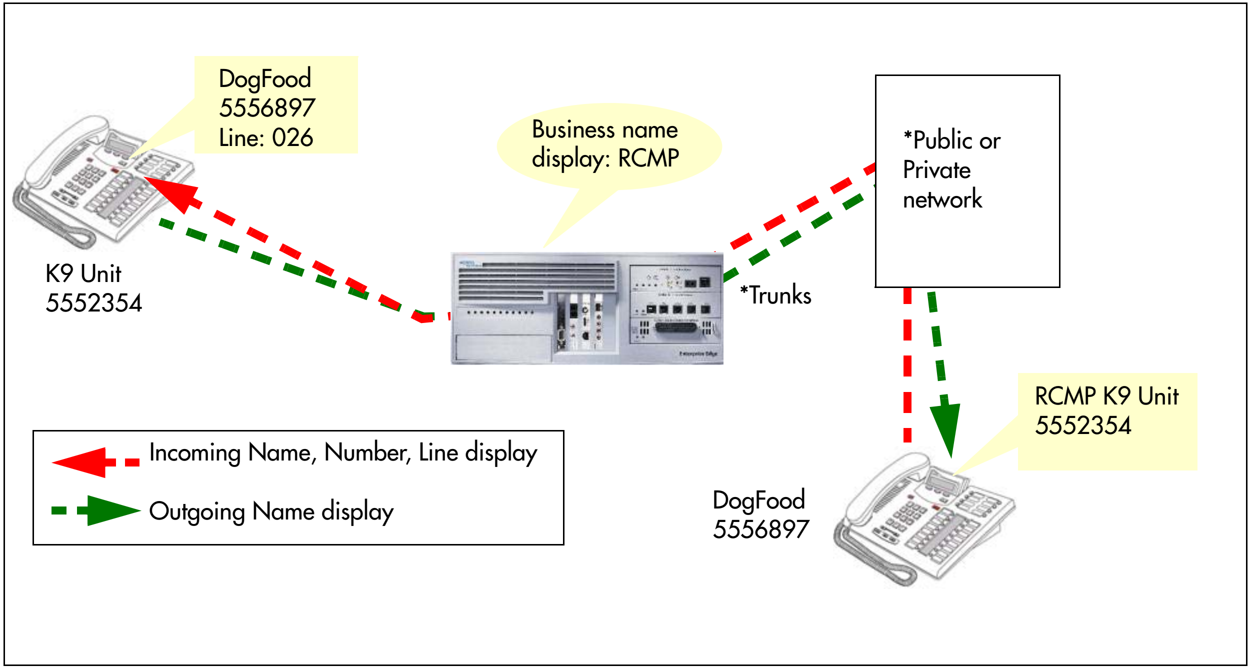
Depending on the services you subscribe to, incoming Call Display information can contain up to three parts:

- the name of the caller
- the number of the caller
- the name of the line in your system that the call is on

Call display information can also be sent out when a system telephone calls out of the system. What displays at the called party’s telephone, depends on what the private or public lines allow. Outgoing call display information can be allowed or blocked at the system level or single telephone level.

Figure 78 illustrates an example of incoming and outgoing call display.

Figure 78 Sending and receiving call display



Chapter 24
Dialing plans

The BCM allows for flexible dialing plans using access codes, destination codes, PSTN trunks and private network trunks that provide multiple options for customizing the dialing options to meet each customers unique requirements. Refer to "Outgoing call routing" on page 252.

While the BCM can be plugged in and used immediately, it is recommended that you plan and execute the appropriate dialing plan.

The dialing plan includes:

- the dialing plans that govern the expected dialing strings on a private network
- the allowed dial strings on a public network
- the access and destination codes that get dialed out as part of the dialing string
- access codes that identify a call type on incoming MCDN calls

These topics are discussed under the following sections:

- "Creating dialing plans" on page 247
- "Public and Private Received numbers" on page 250
- "Private network dialing" on page 251
- "Setting up public network dialing" on page 251
- "Outgoing call routing" on page 252
- "Incoming call routing" on page 254
- "Determining line access dialing" on page 258
- "Understanding access codes" on page 259
- "Line pool access codes" on page 263
- "Using Carrier codes" on page 264
- "Configuring call routing" on page 264
- "Configuring Call-by-Call services" on page 265
- "Using destination codes" on page 269
- "Setting up VoIP trunks for fallback" on page 274

Also refer to the section about call security which deals with defining restriction filters for outgoing calls and remote access packages for incoming calls. This section also discusses Class of Service (CoS) passwords, which can be used when you allow users to access the system features over public connections. Refer to "Call security and remote access" on page 441.

Creating dialing plans

Dialing plans allow users to access the public network, to make calls, and to answer dial strings.

Access to and from and within your system is based on dialing strings and how the system adds or deletes digits from this sequence to route the call.

A dialing string is the numbers that the caller physically enters on a telephone or programs onto a memory key. This can also include numbers the system adds to a dial string when a call goes through call routing.

This process also includes how the receiving system reads the sequence. All of which means that coordination is required at both ends of the call to ensure that calls are routed correctly. This is especially important if calls need to be routed through your system, or through a remote system, to reach another node on the network.

Basic numbering: The first numbering that you set is your DN length (Start DN length) and Start DN and Public and Private Received # length. DN length and Start DN information is entered when the system is initially set up. These numbers can be changed after the system has been set up, but only at the risk of compromising other numbering in the system. If your system is part of a network, these numbers must be coordinated with the other nodes in the network to ensure that the network dialing plans are consistent. The Public and Private Received Number lengths take their sequence from the initial DN length, but this can be changed to accommodate local dialing requirements, the Private length should mirror the DN length, except in special circumstances. Refer to "Incoming call routing" on page 254.

Variable	Example settings
Start DN	2 (221)
DN length, Received # length	
Private length	3
Public length (max)	12 (North America)

Remote access: When you set up lines that do not offer DISA directly on the line, you can determine if remote access prompts with DISA or allows auto answering. This determines the Public/Private Auto DN and Public/Private DISA DN settings, which are set under **Telephony > Dialing Plan > Public Network and Private Network**. These numbers will have the same first number as you specified in the Start DN and be of the same length. Remote callers dial the system public or private access number, and then dial either the Private/Public Auto DN or Private/Public DISA DN, as determined by the line setup.

Variable	Example or default settings
Private Auto DN	2XX
Public Auto DN	2XX
Private DISA DN	2XX
Public DISA DN	2XX

Incoming calls: The Private Dialing Plan provides the special codes that identify the system to calls coming over private PSTN or VoIP trunks. Calls that do not match the private dialing plan information, are not accepted by the system.

Variable	Example or default settings
Private network ID	Number that identifies the system as part of the private network
Location code	UDP networks
Private DN length	DPNSS systems only

Calls coming in over private networks or PRI/BRI termination target lines can be set up for each telephone or group of telephones to which the calls are directed. As with other incoming calls, these calls can have a public or private call type that matches to a public or private received number assigned to a target line.

Variable	Example or default settings
Private received number	<CDP: same as DN of telephone> <UDP: LOC code + DN>
Public received number	<North America: 10 digits XXX-XXX-XXXX, the trailing digits are the DN> <DPNSS: maximum number of digits in local dialing pattern>

Outgoing calls: Other network codes include the information about public dialing codes that you enter. To access the codes select **Configuration > Telephony > Dialing Plan > Public Networks.**

The public dialing plan defines which dialing string prefixes are allowed over the public PSTN lines. By defining these dial strings and the length of the prefix, the central office can direct the calls to the correct public destination.

Variable	Example or default settings
Public DN lengths (prefixes)	Public dialing table

For private networks, if you are not using routing and destination codes, you need to identify an access code that indicates an incoming call is destined for the private network.

Variable	Example or default settings
Private Access Code	6

MCDN special call types: If your system is networked to other types of systems, such as Meridian 1, which sends calls through one or more BCM systems to the public network, you need to specify specific call-type codes. These codes append to the incoming dial string, so that the call-type remains intact as it passes through the BCM call processing:

Variable	Example or default settings	
Local Access Code	9	Coordinate these settings with Meridian routing for these calls types and the Private Access Code.
National Access Code	61	
Special Access Code	911	

Internal feature access: Meanwhile, you need to keep in mind that the leading digit of any of the above dialing codes cannot conflict with the other system access codes that you want to use:

Variable	Example or default settings
Park Prefix	1 (101-125)
Direct Dial Digit	0

Line pool and destination access codes: Once these basic numbers have been picked, you can decide what numbers to use for line pool access codes and/or destination codes. The system will not allow these codes to start with any of the numbers currently assigned. If you are working with an established dialing plan, you may want to ensure that the numbers that the users are familiar with dialing are reserved for these codes.

For instance, if the users are familiar with dialing 9XXXXXXX to access numbers outside of their own offices, you will want to reserve this number for the destination codes. If you are setting up a new system, you could opt to use the location codes of the other systems as destination codes, or you could define one number for local calls (but which are still outside the system) and one number for long distance calls. For example: The users may dial 6<DN number> for calls within a local system, but dial 8<area code><office code><extension or "DN"> for calls in another city over the public network.

Variable	Example or default settings
Line pool codes (first character)	5
Destination codes (first character)	6<up to 11 more characters> 9<up to 11 more characters>

Telephones use pool codes and destination codes to dial externally, because when the analog device goes off hook, it seizes internal dial tone from the system. The external access code, is either a line pool code, or destination code assigned to your system dialing plan.

Variable	Example or default settings
External code	9

Public and Private Received numbers

If the received number is different than the regular DN number, in the target line configuration programming, enter the number in the **Private number** and/or **Public number** field.

Programming note: Auto-answer trunks such as PRI, T1, BRI, and VoIP trunks, use these settings to route calls:

- DPNSS lines use the Private received number to route calls in the system.
- BRI (ETSI-QSIG), PRI (ETSI-QSIG, MCDN, DMS100, DMS250) and VoIP trunks route calls on a per-call basis to either the public or private received digits.

Note: VoIP trunking does not support Auto DN/DISA DN functionality.

- BRI (ETSI-Euro, NI), PRI (ETSI-Euro, NI, 4ESS), T1 (LoopStart, E&M, DID, GroundStart), Analog LEC (LoopStart), and DASS2 trunks route calls using the Public received number.

Private network dialing

If your BCM is part of a private network, you have a choice of dialing plans. However, all BCMs on a network must use the same type of dialing plan and have the same Private DN lengths to ensure proper call direction. Plan out these settings before you start programming for the private network.

- UDP (Universal Dialing Plan) uses a destination code and a location code plus the set DN (that is, 6-403-XXXX) to determine where a call gets routed. You specify a Private DN length to allow all required digits to be dialed. Each node on the network has a unique location code.
- CDP (Coordinated Dialing Plan) uses a unique steering code that is transparent to the user and is dialed as part of the destination set's DN (that is, 2XXXX for one node, 3XXXX for another node, and so on) to determine where the call gets routed. Since each node on the network has a unique code, no other routing is required.
- The Meridian system administrator, or the call control system, generates the Private Network IDs. These IDs are unique to each node on a network. Both UDP and CDP must include this code in programming.

Setting up public network dialing

The public network settings allows you to enter DN lengths for the networks the callers are allowed to dial, including special numbers such as 411 and 911.

The public DN lengths table is used for all PRI calls except for those routes that use service type Private or service type TIE with DN Type specified as Private. This table allows the BCM to determine the length of a DN, based on the initial digits dialed.

A set of default Public DN lengths is included with the default template. In most cases it is not necessary to change the default values.

About the Public DN lengths table

In the public DN lengths table:

- You can define up to 30 entries.
- Each entry consists of a DN prefix string (1 to 10 digits) and a length value (two digits, 1 to 25).
- Several entries are predefined in the North America profile. These defaults can handle most regions in North America without the need for additional programming. If required, you can remove or modify these entries.
- The table always contains one default entry. You cannot remove this entry. You can only modify the length parameter associated with this entry. The default entry specifies the length of any dialing string that does not match one of the other table entries.

Outgoing call routing

Outgoing calls require line pool access codes or destination code (with defined routes) to leave the system.

- Access codes provide direct, unscheduled access to an analog, digital (T1).
- Destination codes also provide access to line pools, but they also allow more flexibility in dialing, which allows for more complex routing options, such as scheduling, fallback routing (VoIP trunks), call definition, and multiple routing (least-cost routing). Routing also allows you to minimize the dialout for the user, especially to systems on the same private network.

Outgoing calls can be either public or private, which is defined by the route. The public or private designation determines which dialing plan is used to determine the validity of the call. Normally, public calls are routed over PSTN trunks and private calls are routed over a private network. However, MCDN trunks can also pass calls designated as public to allow remote nodes on the network to call out of the PSTN of a local node. This is called tandem dialing.

- If the outgoing call is designated as private, the system checks the beginning of the string for a destination code that routes to a private network. It also checks that the dial string is the correct length. The destination code routing determines what the final dial string will be, adding or removing digits, as required.
- If the outgoing call is designated as public, the system checks the beginning of the string for a destination code that routes to a PSTN or an MCDN trunk. If the call routes to a public route, the system checks the public dialing table to ensure that the dialout string has legitimate leading digits and is the correct length. If the call routes to an MCDN trunk, the call is passed as dialed, minus the private networking codes. The call will pass through the system until the system with the matching destination code receives it, at which point it will be sent through the local PSTN of that system.

How the system identifies the call depends on the type of trunk chosen for the route. Refer to the table below.

Dialing plan setting	NPI/TON	Private called number length based on
MCDN trunks send private calls in this way:		
None	Private/Subscriber	Private DN length (set on Private Network panel)

Dialing plan setting	NPI/TON	Private called number length based on
UDP	Private/UDP	private access code + home location code (LOC) + private received digits
CDP	Private/CDP	private received digit
DMS100/DMS250/ETSI-QSIG trunks send private calls in this way:		
None	Private/Subscriber	Private DN length (set on Private Network panel)
UDP	Private/Subscriber	private access code + home location code (LOC) + private received digits
CDP	Private/Subscriber	private received digit

Outgoing public calls routing

Outgoing public calls from within the system typically have the routes set to Public. Refer to "Configuring call routing" on page 264. The NPI/TON is sent as Unknown/Unknown. The public called number length is based on the Public DN lengths table in the Public networks dialing plan.

MCDN trunks also allow public call types when tandeming calls from another system on the private network. Some of these systems use specific call types that the BCM needs to recognize to pass on correctly. Also refer to "Using the MCDN access codes (tandem calls)" on page 262.

Type of call	NPI/TON	BCM prepend access code	BCM monitor display
Local	E164/Local	Local access code (9)	E.164/Subscriber
National	E164/National	National access code (X1)	E.164/National
Special calls (international, 911, etc.)	Private/Special	Special access code (9)	

Incoming call routing

Incoming call routing also depends on the call type. The system also uses the Public and Private DN length settings to determine call routing.

Defining DN length

The DN lengths setting allows you to change the number of digits for the Received number length and the DN length, which are used by the system to determine if an incoming call is valid for the system.

Each increase in length repeats the first digit in front of any existing DN. For example, if DN 234 was increased to a length of four, the new DN would be 2234.

⚠ **Warning:** Do not change DN length immediately after a system start-up. You must wait until the system is operational with two solid green status LEDs.

⚠ **Warning:** Increasing the DN length affects other areas of the system:

If the DN length change creates a conflict with the Park prefix, external line access code, direct-dial digit, or any line pool access code, the setting for the prefix or code changes to None, and the corresponding feature is disabled.

Optional applications affected by DN length changes:

Voice Mail and **Contact Center** applications are reset if you change the DN length after these services are installed.

If you increase your DN length and then decide to decrease the DN length you will have to cold start your system and lose all of the programming.

⚠ **Warning:** If your system is running with a PBX telephony template, the Public and Private received # length are by default 3 (digits) at startup. Increasing the DN length after system startup does not change these digits, so you will need to manually change the Public and Private Receive Number length.

Private OLIs are automatically assigned to the DN records if the DN length and the Private Received Number length are the same. If this changes, the Private OLIs are cleared, or are not assigned (PBX template).

Network note: If your system is part of a private network, ensure that you confirm the dialing plan for the network before changing this length. If you change the length, ensure that you check all DN-related settings after the change.

Using the Received # length

If you change the DN length of your system, you may need to change the Received # length. Private and public networking, and the access codes to determine a route for an incoming call over an auto-answer trunk.

On systems running the DID telephony template, the Private and Public Received # length is set to the same length as the DN length for the system. On systems running the PBX telephony template, the Private and Public Received # length default to 3, unless the DN length is changed during the Startup procedure.

These digits identify target lines ("Processing incoming calls" on page 255), Auto DN*s*, and DISA DN*s*.

The received number can be shorter if network or central office constraints require this. This number cannot be greater than the system DN length on a networked system using a coordinated dialing plan (CDP) or a universal dialing plan (UDP). On a standalone system it is possible that the received number length would be greater than the DN length.

⚠ **Warning:** Decreasing the received number length clears all programmed received digits that are longer than the new settings.

Processing incoming calls

To process an incoming call:

1 The system receives a call from the public or private network.

2 The system identifies the call type:

Public calls:

- If the call is from the MCDN network and is a local, national, or special call type, the system prepends the appropriate access code.

- If the call is from (ETSI-QSIG, MCDN, NI, DMS100, DMS250) and tagged as Private/Subscriber, the system prepends the Private access code, if the dialing plan is UDP.
- If the call is tagged as Unknown/Unknown or Private/Unknown (ETSI-QSIG, MCDN, N1, DMS100, DMS250 trunks), no access code is added.
- For all other call types, the system truncates the trailing digits to the Public Received # Length. (Go to step 4)

Private calls:

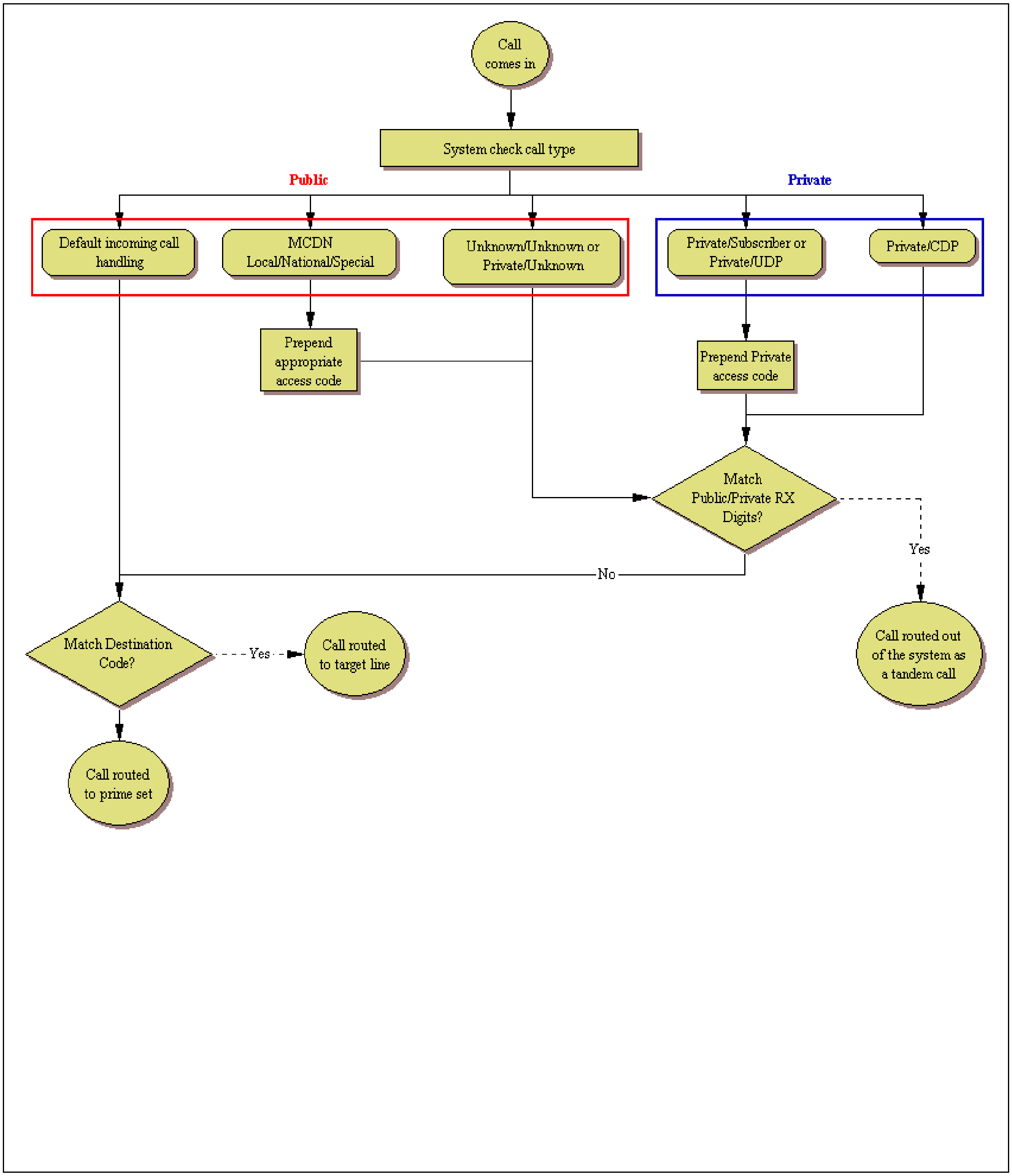
- If the call is tagged as Private/Subscriber or Private/UDP, the system prepends the Private access code.
- If the call is tagged as Private/CDP, no access code is added.

3 The system tries to match the first digits of the dial string to a destination code. If the digits match, the dial string is routed out of the system.

4 If the system cannot match the first digits to a destination code, the system tries to match the dial string to a target line (Public or Private Received Number). If the dial string does not match any target lines, the call is routed to the prime set for the line.

Figure 79 illustrates the incoming call processing.

Figure 79 Incoming public and private call coding


Call comes in
System check call type
Public
Private
Default incoming call handling
MCDN Local/National/Special
Unknown/Unknown or Private/Unknown
Private/Subscriber or Private/UDP
Private/CDP
Prepend appropriate access code
Prepend Private access code
Match Public/Private RX Digits?
Yes
No
Call routed out of the system as a tandem call
Match Destination Code?
Yes
Call routed to target line
Call routed to prime set

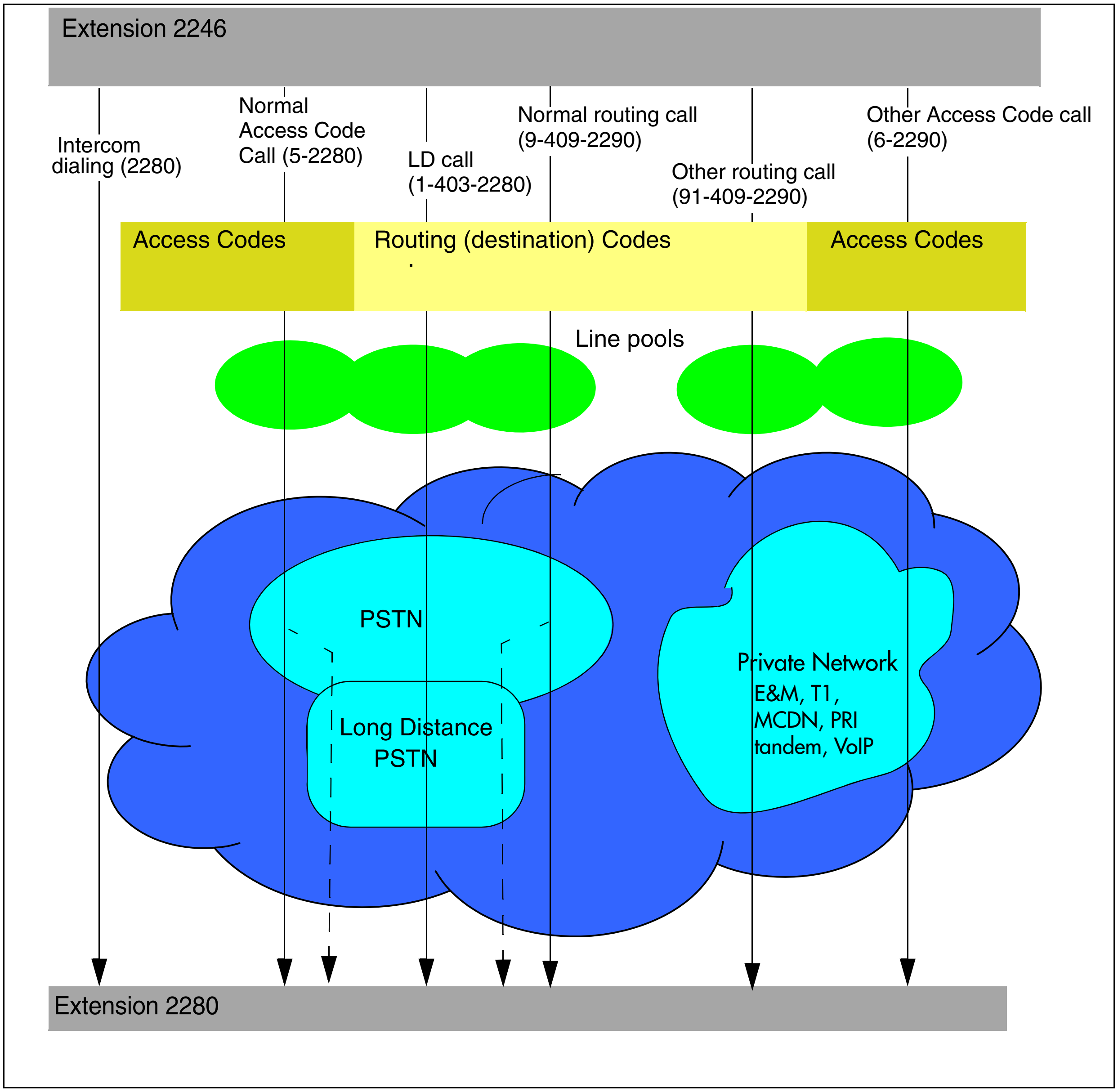
Determining line access dialing

"Understanding access codes" on page 259 and "Configuring call routing" on page 264 describe what you do with the lines and loops you previously set up into line pools.

By using access codes or call routing, which uses destination codes, you can determine which lines (routes) outgoing calls use. When you create a route, you can also specify restrictions that apply to how or when the line will be used.

Figure 80 provides an overview of how access codes and routing is used within the system to direct calls from a telephone in one system to a telephone in another system.

Figure 80 Line management diagram



Understanding access codes

The system uses access codes to direct calls to the correct lines and destinations. Refer to "Creating dialing plans" on page 247 for a general overview about using access codes within the system dialing plan.

Task:

Set up access codes for internal features:

- park prefix
- direct dial digit

Set up access codes that affect users dialing in from remote locations:

- Private Auto DN
- Public Auto DN
- Private DISA DN
- Public DISA DN

Set up access codes that affect calls coming in over the private network:

- Private access code
- Local access code
- National access code
- Special access code

Set up access codes that affect calls leaving the system:

- External code (ATA and analog devices)
- Line pool access codes
- Destination codes
- Carrier codes

The default settings shown in Table 54 can help you plan your access codes so there are no conflicts.

Table 54 Default codes table

Digit	Use	System panel
0	direct dial digit	Configuration > Telephony> Dialing Plan > General > Access codes
1	park prefix	Configuration > Telephony> Dialing Plan > General > Access codes
2XX	first digit of DNs/DN lengths	Set through Startup Profile
9	line pool A access code (Takes precedence over the External line destination code if there is a conflict.)	Routing

Call Park codes

When you park a call (**FEATURE 74**), the system assigns one of 25 codes for the retrieval of the call. You can then press the Page display key to announce the code that appears on the display.

These three-digit codes include the Call Park prefix, which can be any digit from 1 to 9, and a two-digit call number between 01 and 25. For example, if the Call Park prefix is 1, the first parked call is assigned Call Park retrieval code 101.

Note: The Park prefix must not conflict with the following:

- external code
- direct dial digit
- private access code
- Public/Private Auto DN
- Public Private DISA DN
- line pool code/destination code, or
- telephone DN

Note: Other programmable settings may affect which numbers appear in the window during programming. Although the numbers 0 to 9 are valid Park prefix settings, some may already be assigned elsewhere by default or by programming changes.
If the DN length changes, and the changed DNs conflict with the Park prefix, the setting changes to None.

The system assigns Call Park codes to calls in sequence, from the lowest to the highest, until all the codes are used. The use of different of codes ensures a call reaches the right person, especially when more than one incoming call is parked.

→ **Note:** Model 7000 phones are supported in Europe only.

The highest call number (the Call Park prefix followed by 25) is used by model 7000 and 7100 telephones, analog telephones, or devices connected to the system using an ATA2. Analog telephones or devices cannot use the other Call Park codes.

When parking a code on an analog telephone, the call is parked on the highest park code. When retrieving a call, any phone can retrieve the call by entering the park code.

Calls are retrieved by pressing the intercom button and dialing the retrieval code. On model 7000 and analog telephones, pick up the receiver; if the call is parked by the analog phone, use the **<*parkcode*>25,** otherwise, use **<parkcode><parknumber>**.

→ **Note:** Analog phones can park call only at <parkcode>25.

You also need to program the park timeout. The park timeout determines when external parked calls that are not answered return to the originating telephone. See the *BCM 4.0 Device Configuration Guide* (N0060600) for information on programming park timeout.

You can disable Call Park by setting the Park prefix to None.

Creating Direct Dial sets

The Direct dial setting allows you to dial a single system-wide digit to call a specific telephone, called a direct dial telephone. The most common example of a direct dial set is a telephone for an operator, a receptionist or an attendant. You can program a maximum of five direct dial sets on the system, however, you can only specify one direct dial number for the system.

Tips about access codes

Use the following tips to assist you in planning the access codes for your system.

Note: The following codes/digits must not conflict:

- park prefix
- external code
- direct dial digit
- private access code
- Public/Private Auto DN
- Public/Private DISA DN
- line pool code/destination code
- telephone DN

- **External line access code**: If the DN length is changed, and the changed DNs conflict with the external line access code, the setting changes to None.
- **Direct dial telephone:** Another direct dial telephone, an extra dial telephone, can be assigned for each schedule in Services programming.

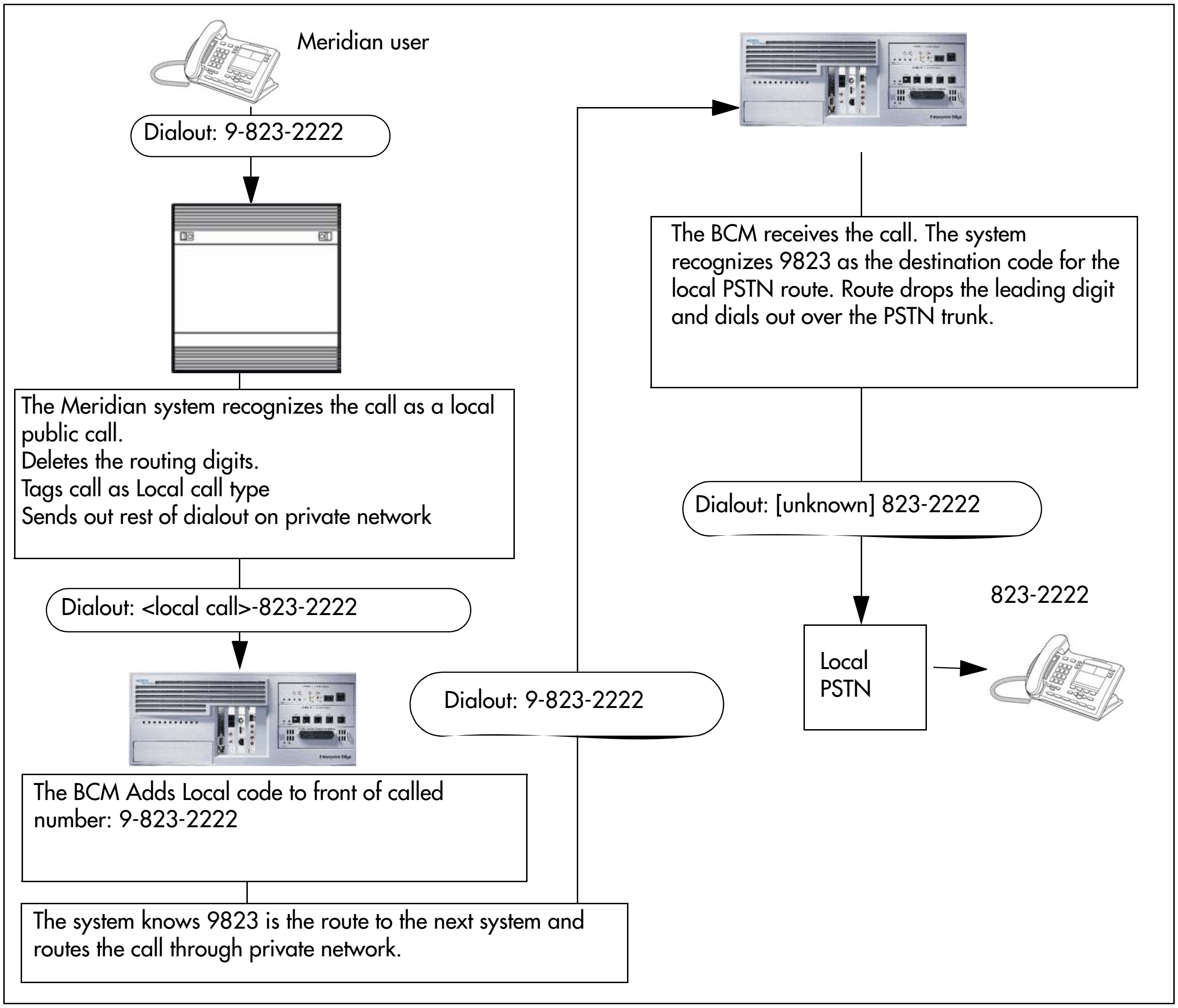
  If the DN length is changed, and the changed DNs conflict with the Direct dial digit, the setting changes to None.
- **Public/Private Auto DN:** The length of the Auto DNs are the same as the Public or Private Received Number Lengths specified under Configuration > Telephony > Dialing Plan. The public/private Auto DN is cleared if the corresponding Received Number Length is changed.
- **Public/Private DISA DN:** The length of the DISA DNs are the same as the Public or Private Received number length specified under Configuration > Telephony > Dialing Plan. The public/private DISA DN is cleared if the corresponding Received number length is changed.

Using the MCDN access codes (tandem calls)

Three special codes exist specifically for programming over PRI and VoIP trunks that are using the MCDN protocol, and which connect to a call server systems that use specific call codes for special call types, such as the Meridian 1 (M1). The purpose of the codes is to allow easier programming of the call server systems when calls are tandemed through a BCM to the local PSTN.

Calls tandeming to the public network through the private network need to retain their dialing protocol throughout the private network. This means that a call from an M1 node tagged as a local call gets received by the BCM node and is recognized as a call intended for the public network, but also as a call that needs to maintain the local call tag until it gets to the BCM node that is directly connected to the PSTN. This is accomplished by ensuring that the destination code, which starts with this access code, passes the call on using the route designated with the correct call type. Figure 81 charts this process.

Figure 81 Local call tandemed through BCM nodes


Meridian user
Dialout: 9-823-2222
The Meridian system recognizes the call as a local public call.
Deletes the routing digits.
Tags call as Local call type
Sends out rest of dialout on private network
Dialout: <local call>-823-2222
The BCM Adds Local code to front of called number: 9-823-2222
The system knows 9823 is the route to the next system and routes the call through private network.
Dialout: 9-823-2222
The BCM receives the call. The system recognizes 9823 as the destination code for the local PSTN route. Route drops the leading digit and dials out over the PSTN trunk.
Dialout: [unknown] 823-2222
823-2222
Local PSTN

Calls coming in from the public network need to be translated to their private network destination before routing/tandeming through the private network. In this case, the route used is defined with the call type of Private.

Line pool access codes

Line pool access codes allow you to assign an access code for each of the basic line pools (A to O). These codes specify the line pool for making an outgoing external call. Up to three digits in length, these codes do not allow any other routing programming. The user simply dials the code in front of the dial string. The system, in turn, deletes the entire code before sending the call out over the appropriate route.

If you need a more complex routing arrangement, you need to specify routes and destination codes, which allows you more flexibility in terms of dial strings, routing schedules, and routing restrictions.

Using Carrier codes

A multi-digit Carrier access code contains an Equal Access Identifier Code (CAC) followed by a Carrier Identification Code (CIC). The CIC identifies the carrier that handles the call. The Carrier Access Code table stores the CAC digit pattern that you define for your region.

In most cases it is not necessary to change the default values.

About Carrier access codes

The following points apply to carrier access codes:

- You can define up to five carrier codes.
- Two entries will be predefined in North America, but you can remove these defaults.
- Each entry consists of an equal access identifier code prefix (one to six digits) and a carrier identification code length (one digit, 1 to 9).
- Each entry is identified by the prefix digits themselves.

Configuring call routing

Call routing allows you to define how calls are routed by your BCM system.

Call routing decides what path an outgoing call takes using the digits that are dialed. It is sometimes called Automatic Route Selection (ARS).

When you select an internal line and dial, the system checks the numbers you enter against the routing tables. If the number you dial starts with a destination code, the system uses the line pool and dials out digits specified by the route assigned to that destination code, and then dials the rest of the number that you dialed.

Routing service replaces a number of manual tasks, including:

- entering a line pool code
- dialing an access code for a long distance carrier
- deciding which line pool to use according to the time and day

You can set up routing to take advantage of any leased or discounted routes using information supplied by the customer. The system cannot tell what lines are cheaper to use.

For Call by Call service selection (PRI only), the installer defines destination codes for various call types over PRI lines (for example, Foreign Exchange, TIE Trunk, or OUTWATS). The user dials a number using the intercom button without entering any special information. For more information see “Provisioning for Call by Call limits with PRI” on page 268.

⚠ **Warning:** Plan your routing service before you do any programming.
Routing affects every call placed in the system and must be carefully planned to avoid conflicts and gaps in the programming. Use tables to design routes and destination codes, then check for potential problems before you start programming. It also saves you time when all the settings are written out in front of you.

Routing configuration

The settings for a call routing include:

- a three-digit route number (000-999)
- external # digits (up to 24 digits)
- a line pool
- destination codes (max. of 500 available, up to 12 digits)
- DN type and/or Service Type
- public and private DN lengths
- a schedule (optional)

Configuring Call-by-Call services

Call-by-Call service selection (CbC) allows you to access services or private facilities over a PRI line without the need for dedicated facilities. The different services represent different types of access to the network.

This section includes information about:

- "Call-by-Call services" on page 266
- "Switches supporting Call by Call limits" on page 266
- "Provisioning for Call by Call limits with PRI" on page 268

Supporting protocols

The following protocols support Call by Call limits:

- National ISDN 2 (NI-2)
- DMS-100 custom
- DMS-250
- AT&T 4ESS custom

Call-by-Call services

BCM supports the Call-by-Call Services listed in Table 55.

Table 55 Call-by-Call Services available on the system

Service	Description
Public	Public calls connect BCM and a Central Office (CO). BCM supports both incoming and outgoing calls over the public network. Dialed digits conform to the standard North American dialing plan (E.164 standard).
Foreign Exchange (FX)	Foreign exchange service connects a BCM site to a remote central office (CO). This provides the equivalent of local service at the remote location.
TIE	TIE lines are private incoming and outgoing lines that connect Private Branch Exchanges (PBXs) such as another BCM.
OUTWATS	Outward Wide Area Telecommunications: This outgoing call service allows a BCM user to call telephones in a specific geographical area referred to as a zone or band. Typically, a flat monthly fee is charged for this service.
INWATS	Inward Wide Area Telecommunications: This long distance service allows a BCM user to receive calls originating from specified areas without charge to the caller. A toll-free number is assigned to permit reverse billing.
International INWATS	An international long distance service that allows a BCM user to receive international calls originating from specified areas without charge to the caller. A toll-free number is assigned to permit reverse billing.
Switched Digital	This service provides premises-to-premises voice and data transport with call management and monitoring features.
Nine Hundred	This service is commonly referred to as fixed-charge dialing.
Private	Private incoming and outgoing calls connect BCM to a virtual private network. Dialed digits can conform to the standard North American dialing plan (E.164 standard) or the dialed digits can use a private dialing plan.

Switches supporting Call by Call limits

Table 56 lists the service types and cross-references them with four common switches.

Table 56 Switches and service types chart (Sheet 1 of 2)

	Switches			
Service types[1]	**NI-2**	**DMS-100 (custom)**	**DMS-250**	**AT&T 4ESS**
FX	FX	FX[2]	N/A	N/A
Tie[3]	TIE	TIE	TIE	SDN (software defined network)
INWATS	INWATS	INWATS	Eight Hundred	Toll Free MEGACOM
International INWATS	Same as INWATS	Same as INWATS	Same as INWATS	International Toll Free Service
OUTWATS	IntraLATA OUTWATS OUTWATS with bands InterLATA OUTWATS	OUTWATS	PRISM	MEGACOM
Private		DMS Private[5]	VNET (virtual network)	N/A

Table 56 Switches and service types chart (Sheet 2 of 2)

	Switches			
Service types[1]	**NI-2**	**DMS-100 (custom)**	**DMS-250**	**AT&T 4ESS**
Switched Digital	N/A	N/A	N/A	ACCUNET[4]
Nine Hundred	N/A	N/A	Nine Hundred	MultiQuest
Public	Public	Public	Public	N/A

1. N/A indicates that the protocol does not support the service.
2. DMS-250 Sprint and UCS support incoming FX only (that is, Network-to-BCM).
DMS-250 MCI does not support FX.
3. NI-2 allows two TIE operating modes: senderized and cut-through. BCM supports only senderized mode.
4. Rates greater than 64 kbps are not supported.
5. Bell Canada VNET.
6. Not all service types may be supported by a switch type. For information, contact your service provider.

Provisioning for Call by Call limits with PRI

To program the system for Call by Call Limits with a PRI interface, you must:

- provision a DTM as PRI, if one is not already configured as part of the system
- select a protocol
- program incoming call routing
- program routes that use the PRI pools, see "Configuring call routing" on page 264.

Other required programming in the Element Manager

Programming Call by Call on PRI requires these settings:

- Select **Configuration > Sets > All DNs**, and assign the line pool.
- Select **Configuration > Telephony > Dialing Plan > Routing**, assign a pool for routing, and assign the service type and service id, if required.
- Select **Configuration > Telephony > Dialing Plan > General,** and specify the minimum and maximum values for the pools.

Call by Call service routing

Table 57 is an example of a Routing Table containing Call by Call programming (available in the North America market profile). Also refer to "Configuring Call-by-Call services" on page 265.

Table 57 Call by Call routing table example

Route Number (000-999)	Dial Out (24 digits)	Use Pool	Service Type	Service Identifier
003		BlocA	Public	
004		BlocA	FX	xxxxx
005		BlocA	TIE	xxxxx
006		BlocB	OUTWATS	xxx
007		BlocB	Private	
008		BlocB	Switched Digital	

Note: The public DN lengths are used for all PRI calls except those whose routes use service type Private or service type TIE with DN Type specified as Private.

➡ **Note:** This type of routing only applies to those PRI trunks set with a protocol of NI, DMS100, DMS250 or 4ESS.

The service identifier (SID) depends on the selected service type (for example, with NI-2 protocol).

Service Type	Service Identifier description
Public	None
FX	Facility Number 1-5 digits
TIE	Facility Number 1-5 digits
OUTWATS[a]	Optional Band Number 1-3 digits
Private	None
Switched Digital	None

[a] For NI-2, do not program the Carrier Access Code for banded OUTWAT calls. This call may be rejected.

When you select or change a PRI protocol, the Service Type and Service ID fields automatically clear for each entry in the routing table for that PRI.

PRI routing protocols

Table 58 lists the service/DN type choices available for PRI lines.

Table 58 PRI Service type/DN type values

PRI Protocol	Type	Values
MCDN	DN	Public, Private, Local, National, Special
ETSI Euro	DN	None, Overlap
ETSI QSIG	N/A	
NI	Service	Public, TIE, Foreign Exchange (FX), OUTWATS
DMS100	Service	Public, Private, TIE, Foreign Exchange (FX), OUTWATS
DMS250	Service	Public, Private, TIE, Foreign Exchange (FX), OUTWATS
4ESS	Service	TIE, OUTWATS, Switched Digital (SDS)

Using destination codes

Destination codes allow you to control how the system interprets and routes dial strings from internal sources. Destination codes are similar to line pool codes except that by using routes (which attach dial strings and DN type designators to line pools) and schedules you can control what digits the user has to dial and how the system routes the call out of the system, including what numbers from the dial string get added or deleted to the route dialout.

→ **Note:** Destination codes must not conflict with the following:

- park prefix
- external code
- direct dial digit
- Auto DN
- DISA DN
- Private access code
- line pool codes
- telephone DN
- public target line received digits
- other destination codes

→ **Note:** You can enter destination codes up to a maximum of 12 digits.

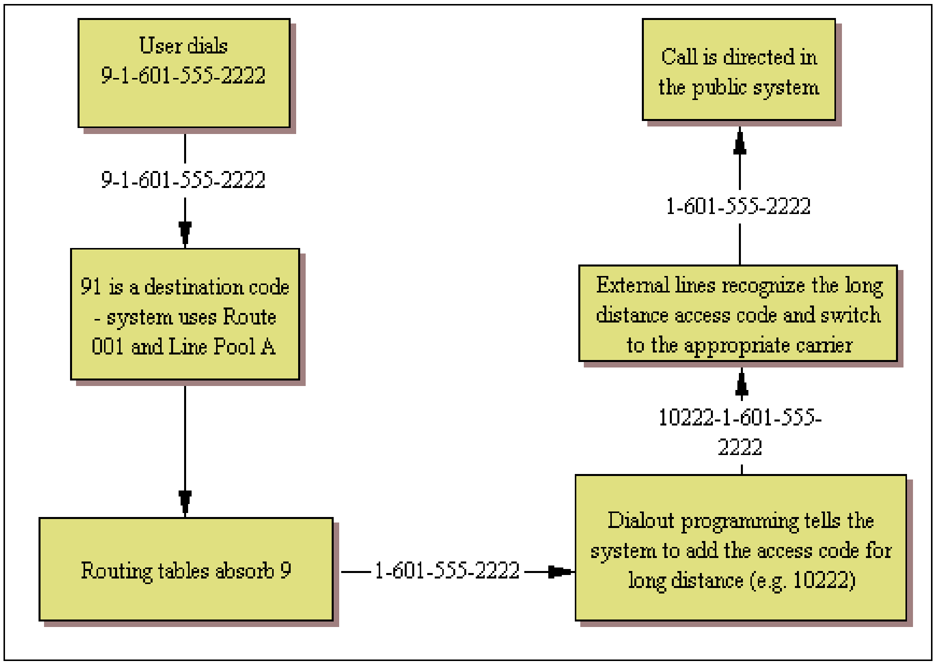
Why use destination codes?

Routes determine path (line or pool) and any required access numbers.

Destination codes determine which route to take (that is, an end node uses one destination code for all other nodes in the system). If you choose to use the destination codes Normal schedule, the call will always go out over the same route. If you choose to use the other destination codes schedules, you can set up a more responsive plan, whereby calls can go out over more than one route, based on scheduled times.

Destination codes provide you with the opportunity to create a dialing plan that allows users to connect to other systems in a relatively seamless or consistent manner, regardless of the lines or routes that are being used to get there. For example, connecting through VoIP lines requires significantly different ways of dialing than dialing over T1 lines. However, you can configure destination codes, such that the user dials the same number of digits regardless of the trunks over which the calls are routed.

Figure 82 Using destination codes to access another system


User dials
9-1-601-555-2222
9-1-601-555-2222
91 is a destination code - system uses Route 001 and Line Pool A
Routing tables absorb 9
1-601-555-2222
Dialout programming tells the system to add the access code for long distance (e.g. 10222)
10222-1-601-555-2222
External lines recognize the long distance access code and switch to the appropriate carrier
1-601-555-2222
Call is directed in the public system

Deciding on a code

When deciding on which digits to use to start your destination codes, consider the following:

- Ensure that the digit or digits you want to start your destination codes with do not match any of the access codes, including the line pool codes that already exist in your system.
  You may find that you need to delete line pool codes and create a route and destination code instead. This could occur if you want to set up fallback to a public line, for instance. If the public line is accessed by a line pool code, you would have to change access to a route so you could create a fallback schedule with the destination code used for the primary line (or lines, if you have more than one outgoing line pool that requires fallback).
- Decide how much of the common part of a dial string you want your users to have to dial, and how much you can put in the dial string.
- If you want specific dial strings to use specific routes, map these out first.

  For instance, if you want users to dial between BCMs over VoIP lines, you would create destination codes specific to those systems which use the VoIP line pool, using the digits with which the users are familiar. You can then create a unique destination code for the call you want to route.

  Example: If users are used to dialing 9-1-555-1234-<DN number> to reach another system (whose DN codes start with 6), you create a destination code of 915551236A, using the VoIP line pools (users dial the destination code plus the DN of the telephone they want to reach on the other system). The letter A at the end of the code represents any number from 0 to 9 which is not used by any other destination code.

If you need to use PSTN lines for a specific connection on the other system, you can create a destination code specific to that destination number and attach it to the route set up with the PSTN line pool (for example, 915551236333, 6333 being the DN of the device on the other system. When the user dials that specific number, the call will always go over the PSTN line). Note that by entering this code, users dialing with the code in the previous paragraph could never dial any DN that started with 63XX.

- If you want to use VoIP lines as your main lines, but you want to program one or more PSTN lines as fallback lines, you need to configure the routing and routing schedules so that the user dials the same number, regardless of which routes get used. You use the external number dialout string and absorb digits fields under the schedules in Destination code programming for this purpose.
- If a company wants to use VoIP lines between sites for interoffice calls, but not necessarily for all the voice traffic, they can configure specific destination codes for the VoIP routes. In this case, the destination code contains the same digits as a user would dial for a PSTN line, thus, making the shift transparent to the user and, at the same time, ensuring that the most economical route is being used. Depending on how many exceptions there are, you can use the wild card at the end of the string to save yourself from the necessity of entering a number of destination codes with the same leading digits.

Configuring Absorbed length

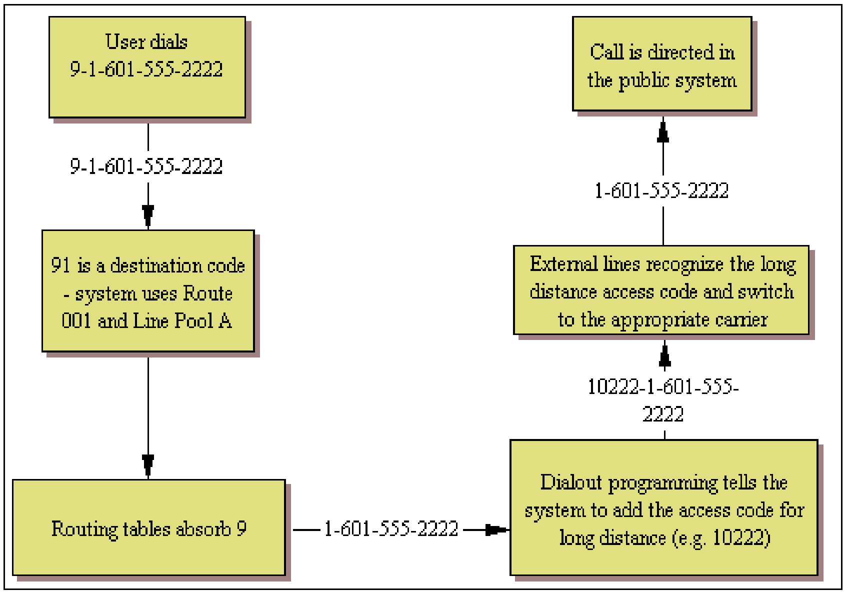
The digit absorption setting (**Absorbed Length**) applies only to the destination code digits.

When the Absorbed Length is at 0, the actual digits dialed by a caller are preserved in the dialout sequence. As you increase the absorbed length the equivalent number of digits are removed from the beginning of the destination code.

Adding Carrier access codes to destination codes

In some instances, long distance service uses the same lines as local service but is switched to a specific carrier using an access number, which is sometimes referred to as a carrier access code (CAC). Route programming can include the access number so the users do not have to dial it every time they make a long distance call. Figure 83 shows an example of how the system interprets what the user dials into a valid outgoing call.

Figure 83 Carrier code call numbering sequence


User dials
9-1-601-555-2222
9-1-601-555-2222
91 is a destination code - system uses Route 001 and Line Pool A
Routing tables absorb 9
1-601-555-2222
Dialout programming tells the system to add the access code for long distance (e.g. 10222)
10222-1-601-555-2222
External lines recognize the long distance access code and switch to the appropriate carrier
1-601-555-2222
Call is directed in the public system

Tips: The destination codes 9 and 91 used in the examples cannot be used together. If you need the destination code 91 to direct long distance calls, you must create a separate set of codes that use local calling routes. These codes would be, for example, 90, 92, 93, 94, 95, 96, 97, 98 and 99. You can also use 9 *A*. (*A* represents a wildcard “Any”.*)*

Routing schedules and alternate routes

It can be less expensive to use another long distance carrier at a different time of day. Continuing with the example used in the previous flowchart, the lines that supply local service in normal mode are also used for long distance service after 6 p.m. because that is when rates become competitive. For the system to do this automatically, you must build another route.

All the lines used by a route specified by a destination code are busy when a call is made, you can program other routes that the system automatically flows the calls to, or you can allow the call to overflow directly to the Normal route schedule (usually the most expensive route). However, this only takes effect if an active routing schedule is applied to the line. Overflow routing is not available in Normal mode.

You must create overflow routes for each destination code for which you want to allow overflow routing.

When a user dials, and the telephone cannot capture the preferred line (First Route), the system tries each successive defined route (Second Route, then Third Route). If none of these routes have available lines, the call reverts to the Normal mode. When the call switches from the preferred routing mode (First Route, Second Route, Third Route) to Normal mode, the telephone display flashes an “expensive route” warning. VoIP trunking uses a similar process for setting up fallback from the VoIP trunk to a PSTN line.

➡ **Note:** Overflow routing directs calls using alternate line pools. A call can be affected by different line filters when it is handled by overflow routing.

Setting up VoIP trunks for fallback

Fallback is a feature that allows a call to progress when a VoIP trunk is unavailable or is not providing adequate quality of service (QoS).

Refer to “Setting up VoIP trunks for fallback” on page 423 for details about setting up fallback for VoIP trunks.

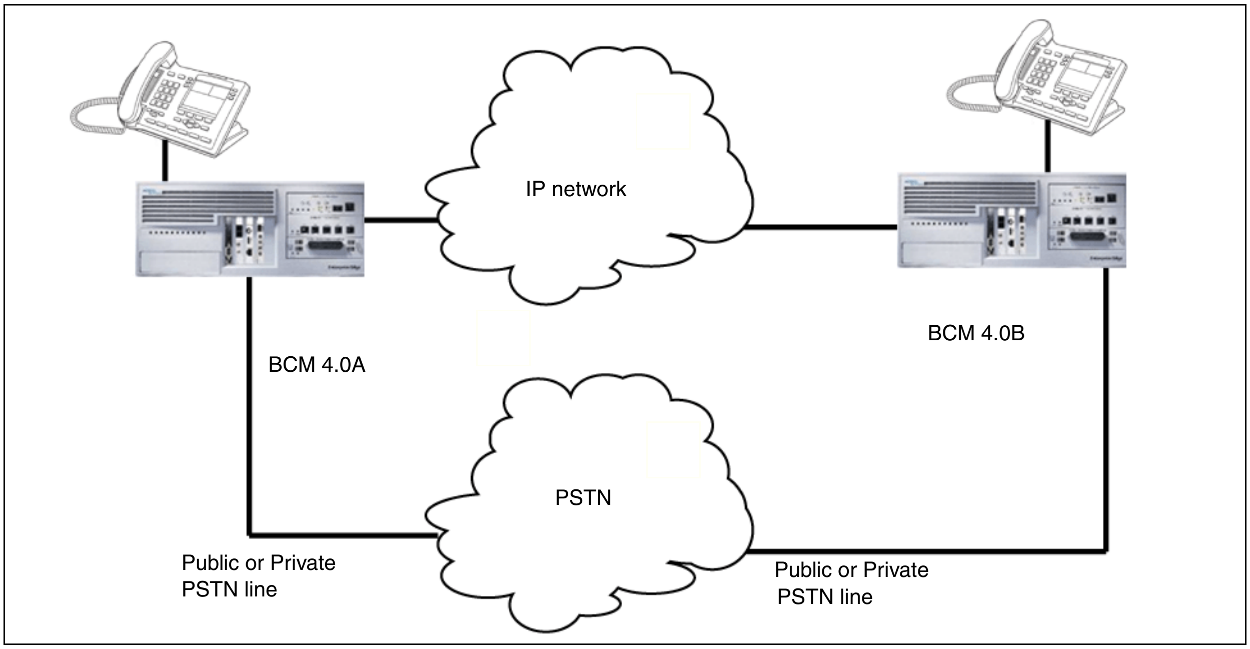
By enabling **PSTN fallback** on the Local Gateway IP Interface panels for H.323 trunks, you allow the call to switch to a PSTN line if the VoIP trunk is not available or cannot produce the expected quality. To access the Local Gateway IP Interface panel, select **Configuration > Resources > Telephony Resources > IP Trunks**.

You use scheduling and destination codes to allow the call to switch from H.323 line pools to a PSTN line without requiring intervention by the user.

Use the dialing plan worksheet in the Programming Records to plan your dialing requirements so you can pinpoint any dialing issues before you start programming. If you are programming an existing system, you can look at what numbers the users are familiar with dialing, and you can attempt to accommodate this familiarity into your destination codes plan.

Figure 84 shows how a fallback network would be set up between two sites.

Figure 84 PSTN fallback diagram



In a network configured for PSTN fallback, there are two connections between a BCM and a remote system.

- One connection is a VoIP trunk connection through the IP network.
- The fallback line is a PSTN line, which can be the public lines or a dedicated T1, BRI, PRI or analog line, to the far-end system.

When a user dials the destination code, the system checks first to see if the connection between the two systems can support an appropriate level of QoS. If it can, the call proceeds as normal over the VoIP trunk. If the minimum acceptable level of QoS is not met, the call is routed over the second route, through the PSTN line.

For PSTN fallback to work, you must ensure that the digits the user dials will be the same regardless of whether the call is going over the VoIP trunk or the PSTN. In many cases, this involves configuring the system to add and/or absorb digits.

For information on how to configure VoIP trunks for fallback, refer to "Setting up VoIP trunks for fallback" on page 423

Chapter 25

Dialing plan: Routing configurations

The following describes how to configure the lines and loops to allow system users to dial out of the system over a public or private network.

The following paths indicate where to access the route lines and loops in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > Routing**
- Telset interface: ****CONFIG > System Programming**

Task: Set up routing for various call scenarios:

- "Destination code numbering in a network" on page 279
- "Setting up a destination for local calling" on page 279
- "Setting up a route through a dedicated trunk" on page 280
- "Grouping destination codes using a wild card" on page 281
- "Programming for least-cost routing" on page 282
- "Using multiple routes and overflow routing" on page 282
- "Using the MCDN access codes to tandem calls" on page 287

Prerequisites

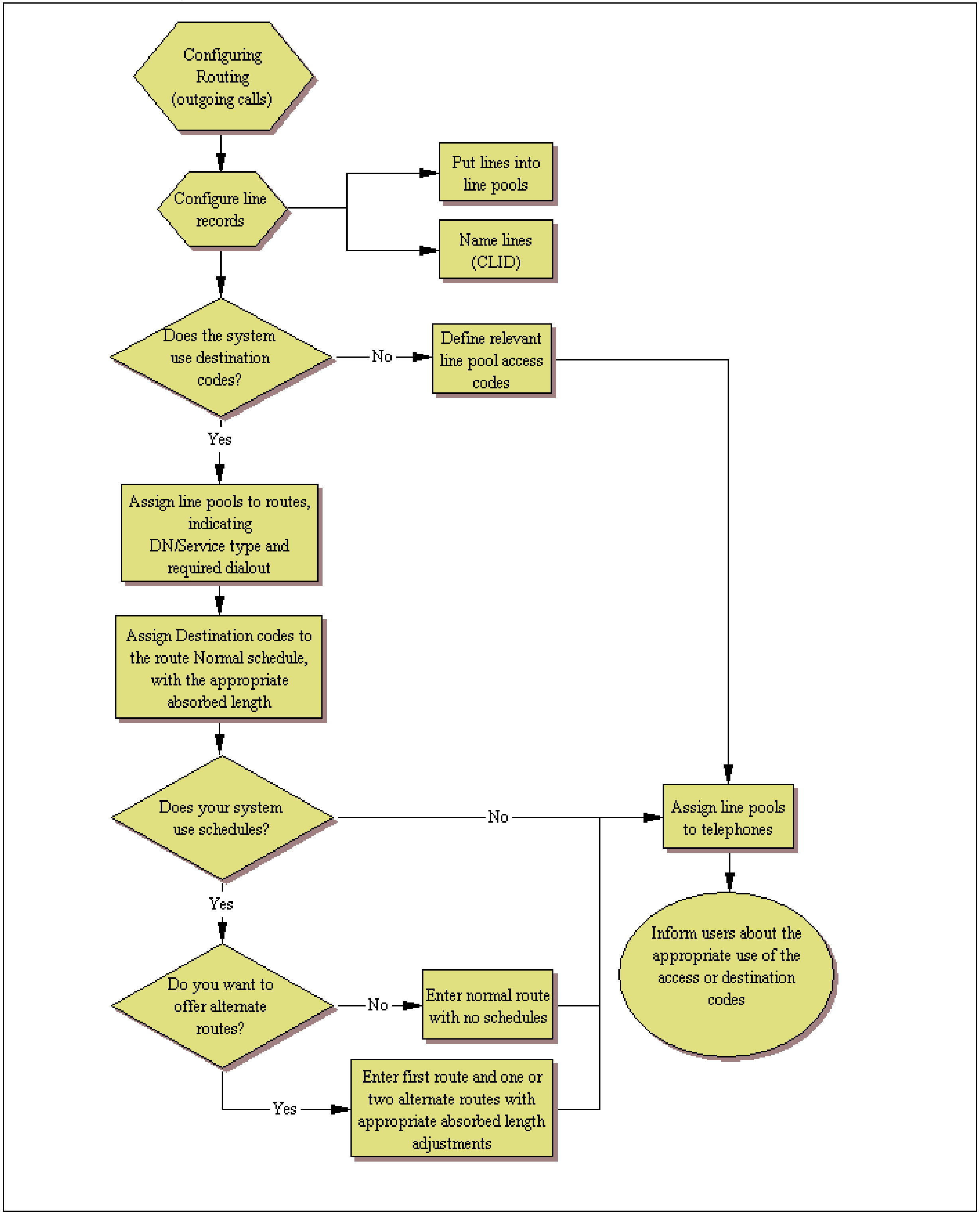
Complete the following prerequisites checklist before configuring the modules.

Media bay modules/VoIP trunks are installed and configured.	
Create an access code/route map to understand how the numbering works for the system.	

Routing work flow

Figure 85 shows an overview task flow for the areas in programming that affect how routes are set up.

Figure 85 Routing workflow


Configuring Routing (outgoing calls)
Configure line records
Put lines into line pools
Name lines (CLID)
Does the system use destination codes?
No
Define relevant line pool access codes
Yes
Assign line pools to routes, indicating DN/Service type and required dialout
Assign Destination codes to the route Normal schedule, with the appropriate absorbed length
Does your system use schedules?
No
Assign line pools to telephones
Yes
Inform users about the appropriate use of the access or destination codes
Do you want to offer alternate routes?
No
Enter normal route with no schedules
Yes
Enter first route and one or two alternate routes with appropriate absorbed length adjustments

Destination code numbering in a network

Because the system checks the initial digits of a call against the routing tables, each type of internal or external call must begin with a unique pattern of digits. Table 59 gives a sample plan for how initial digits are assigned in a network of systems with three-digit intercom numbers.

Table 59 Destination code leading digits

Leading Digits	Use
0	Network Direct Dial
221-253	Intercom calls
4	Coordinated Dialing Plan
5	Unused
6	Unused
1	Call Park Prefix
9	All PSTN Calls
7	Unused

In Table 59, 4 is used as the initial digit for the coordinated dialing plan, but 5, or 6 can also be used for this purpose.

Tips: When programming a button to dial an external number automatically (autodial), private network calls must be programmed as external autodial numbers, even though they resemble internal extension numbers.
Routes generally define the path between the BCM system and another switch in the network, not other individual telephones on that switch.

Setting up a destination for local calling

An office can have different suppliers for local and long distance telephone service. By programming a destination code, any call that begins with 9, which is the most common dial-out digit, automatically uses lines dedicated to local service.

To build a route to allow local calls

1. Create a route that uses the line pool you assigned for the PSTN trunks. Refer to "Routes" on page 290.
2. Create a destination code record and enter a destination code, such as 9, which is a common local call code. Refer to "Grouping destination codes using a wild card" on page 281.

   For local calls only, there are no dial out numbers. Compare with "Setting up a route through a dedicated trunk" on page 280.

   The destination code can use a different route, depending on what schedule is assigned. In the current example, the route you define is used when someone dials 9 during Normal mode, when the other Schedules are turned off.

3 Set up the Normal schedule with the route number you defined in step 1.

Figure 86 Routing Service programming example

Routing Service (Services: Routing Service)		
Route # (000-999)	**Dial out (if required) (max. 24 digits or characters)**	**Use Pool**
001	none	[✓ A] B C D E F G H I J K L M N O
002	none	A [✓ B] C D E F G H I J K L M N O

[Figure 87](#) shows an example of a destination codes programming record filled out.

Figure 87 Destination codes for call routing

Destination codes (Services; Routing service; Destination codes)								
Service Schedule (max. 7 char)	**Normal Rte**		**Route schedule**					
DestCode (max. 7 digits)	**Use route (000-999)**	**Absorb Length**	**1st route (000-999)**	**Absorb Length**	**2nd route (000-999)**	**Absorb Length**	**3rd route (000-999)**	**Absorb Length**
9	003	All						
1	002	0						

An office can have leased lines or private network trunks that provide cheaper to long distance calls by routing through the dedicated lines to remote systems, then using the local PSTN from that system to make the call. The routing should take place automatically when the number of the outgoing call begins with 1.

Setting up a route through a dedicated trunk

If your long distance is supplied by an alternate service or if you want to use different trunks at different times of the day, you can configure a route to use a specific trunk.

To set up a route through a dedicated trunk

1 Create a route that uses the line pool containing the discounted lines for long distance calling. Refer to "Routes" on page 290.

2 Create a destination code record and enter a valid destination code (maximum of 12 digits). Refer to "Grouping destination codes using a wild card" on page 281.

You must use a valid destination code, such as 91 (9, indicating PSTN; 1, indicating a long distance). View existing destination codes before entering a new code. The destination code can use a different route depending on the Schedule.

3 Under the **Normal** schedule for the destination code, enter the route you specified in step 1.

Grouping destination codes using a wild card

If you have a number of destinations that have the same route and digit absorb length, you can group these codes under one destination code to maximize your destination code table. In this case, the start digits will be the same, but the last character will be the wild card, and indicates any digit between 0 and 9. However, if there is a conflict with other digits already programmed or used by other destination codes, an error message appears.

For instance, you might use the same route (555) to a number of remote sites. Each site is accessed with the same external # (dial out string), except for the last digit, which is unique to each site. The exception to this is a site with a totally different access number and line pool requirement (route 565). This example is shown in Table 60.

Table 60 Establishing routes and dialout requirements

Route	Dial Out (external #)	Line Pool
555	0162 237 625<unique number from 0 to 9>	Line Pool C
565	0173 133 2211	Line Pool A

If you do not use wild cards, you would need to create a separate destination code for each unique dialout, as shown in Table 61.

Table 61 Destination codes not using a wild card

Destination codes	Route	Absorb Length	Dial Out
5621	555	3	0162 237 6251
5622	555	3	0162 237 6252
5623	555	3	0162 237 6253
5624	555	3	0162 237 6254
5625	555	3	0162 237 6255
5626	555	3	0162 237 6256
5627	565	All	0173 133 2211
5628	555	3	0162 237 6258
5629	555	3	0162 237 6259

If you use the wild card character *A* (ANY), you can reduce the number of destination codes you require to two, as shown in Table 62.

Table 62 Destination codes using the ANY character

Destination codes	Route	Absorb Length	Dial Out
562A	555	3	0162 237 625X where X is the last digit of the destination code dialed out, from 1 to 9, but not 7
5627	565	All	0173 133 2211

➡ **Tips:** To minimize the effort involved in preparing destination codes, set the digit absorption to 0. When digital absorption is set to 0, the actual digits dialed by a caller are preserved in the dial-out sequence. The need to program a dial out sequence as part of the route depends on the required dialout.

Programming for least-cost routing

It can be less expensive to use another long distance carrier at a different time of day. Continuing with the example used in Figure 85, the lines that supply local service in normal mode are also used for long distance service after 6 p.m. because that is when rates become competitive. For the system to do this automatically, you must build another route.

To build a route for a secondary carrier

1 Create a route for the trunks and assign it to the Normal schedule. Refer to "Setting up a route through a dedicated trunk" on page 280.

2 Choose **No number** for the dial-out.

3 Choose the line pool that contains the local service carrier lines.

4 Now you need to create a destination code and assign the route to the Night schedule. In this case, the change in route uses the start and stop times for Night Schedule.

5 Create 91 as a **Destination code**.

6 Make sure **Absorbed length** is set at 1.

7 Under **Night schedule:** enter the route you defined in step 1.

Calls that begin with the digits 91 travel out without using the access code when the Night schedule becomes active or when you turn it on at a control telephone.

Using multiple routes and overflow routing

If all the lines used by a route specified by a destination code are busy when a call is made, you can program other routes that the system automatically flows the calls to, or you can allow the call to overflow directly to the Normal route schedule (usually the most expensive route). However, this only takes effect if an active schedule is applied to the line. Overflow routing is not available in Normal mode.

You must create overflow routes for each destination code for which you want to allow overflow routing.

To set up the multiple routing overflow feature

1. You assign the preferred routes in a destination code schedule. Refer to "Alternate routes for routing schedules" on page 294.
   - a Pick a schedule when you want these routes to be in effect.
   - b In the **First Route** field enter the route number for the preferred route for the call.
   - c Choose the absorb length for the first route that is appropriate for the dialout numbers you entered for the route.
   - d Repeat steps b and c for **Second Route** and **Third Route** fields.
   - e Define the start/stop time as 0100 under the equivalent Routing Services schedule. This setting means that the schedule is active 24 hours a day. Refer to "Configuring schedule names and timers" in the *BCM 4.0 Device Configuration Guide* (N0060600).
2. Assign an overflow route, usually the most expensive route, to the same Destination Code, but for the Normal schedule. Refer to "Destination codes" on page 292.
3. On the Scheduled Services table, choose auto for Service Setting, and enable Overflow. Refer to "Configuring scheduled service" in the *BCM 4.0 Device Configuration Guide* (N0060600).
4. Use a control telephone to activate or override the feature on the telephones on which you want preferred routing to be active.

➡ **Note:** You must also ensure that the route correctly absorbs or passes dialed digits so that the number dialed for each line is the same from the user perspective.

When a user dials, and the telephone cannot access the preferred line (First Route), the system tries each successive defined route (Second Route, then Third Route). If none of these routes have available lines, the call reverts to the Normal mode. When the call switches from the preferred routing mode (First Route, Second Route, Third Route) to Normal mode, the telephone display flashes an "expensive route" warning.

➡ **Note:** Overflow routing directs calls using alternate line pools. A call can be affected by different line filters when it is handled by overflow routing.

VoIP trunking uses a similar process for setting up fallback from the VoIP trunk to a PSTN line.

This section deals with applying the programming in network situations.

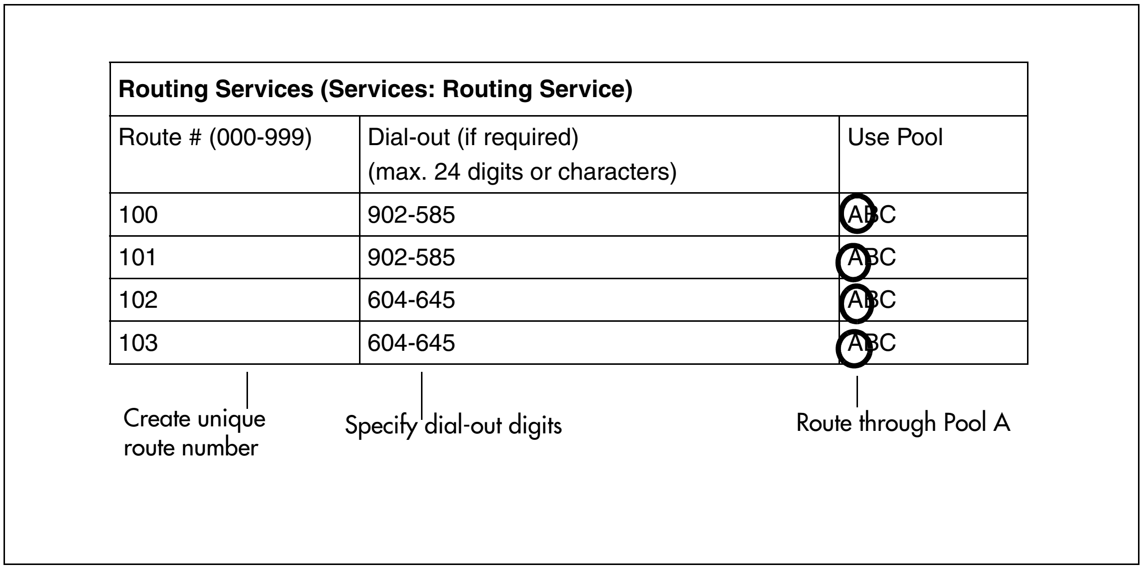
- "Dialing plan using public lines" on page 284
- "Destination code numbering in a network" on page 279

Dialing plan using public lines

Figure 88 and Figure 89 provide examples of how you can record dialing plan information in a spreadsheet. The example shows dialing plan information for a Toronto system in a network of three offices: Toronto, Halifax and Vancouver. Without routing, a BCM user in Toronto would have to select a line pool and dial 1-902-585-3027 to reach extension 27 in Halifax (902). By creating a destination code of 30 and creating a route that uses the proper line pool and dial out number, the user simply dials 3027. The same feature is available for Vancouver (604).

In the column Dial-out, P stands for pause, a host system signaling option. Press **FEATURE 78** to insert a 1.5-second pause in the dialing string.

Figure 88 Routing service record: use pool

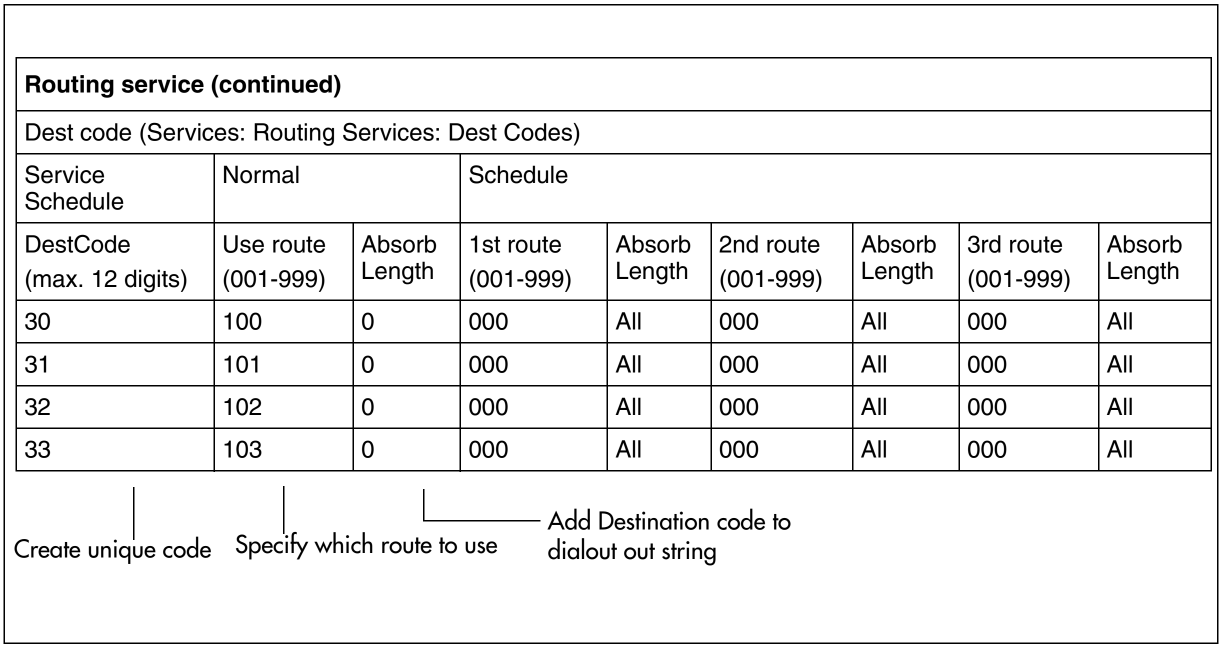


Routing Services (Services: Routing Service)

Route # (000-999)	Dial-out (if required) (max. 24 digits or characters)	Use Pool
100	902-585	[✓ A] B C
101	902-585	[✓ A] B C
102	604-645	[✓ A] B C
103	604-645	[✓ A] B C

Create unique route number
Specify dial-out digits
Route through Pool A

Figure 89 Routing service record: Destination code



Routing service (continued)								
Dest code (Services: Routing Services: Dest Codes)								
Service Schedule	Normal		Schedule					
DestCode (max. 12 digits)	Use route (001-999)	Absorb Length	1st route (001-999)	Absorb Length	2nd route (001-999)	Absorb Length	3rd route (001-999)	Absorb Length
30	100	0	000	All	000	All	000	All
31	101	0	000	All	000	All	000	All
32	102	0	000	All	000	All	000	All
33	103	0	000	All	000	All	000	All

Create unique code
Specify which route to use
Add Destination code to dialout out string

Programming the PRI routing table

The dialing plan must be thoroughly planned out in advance before programming the information into the BCM system.

To program the PRI routing table

1 Click **Configuration > Telephony > Dialing Plan > Routing**.

2 Click the route number record you want to use.

3 In the External Number column, type a dialout number (up to 24 digits).

4 Under Use Pool**,** select a PRI line pool.

The Bloc pools that are displayed depend on how you allocate PRI lines into pools in the line programming. It is possible to have only pool BlocA, or only pool BlocB, even if there are two DTMs configured as PRI in the system.

5 Choose a Service Type or DN type:

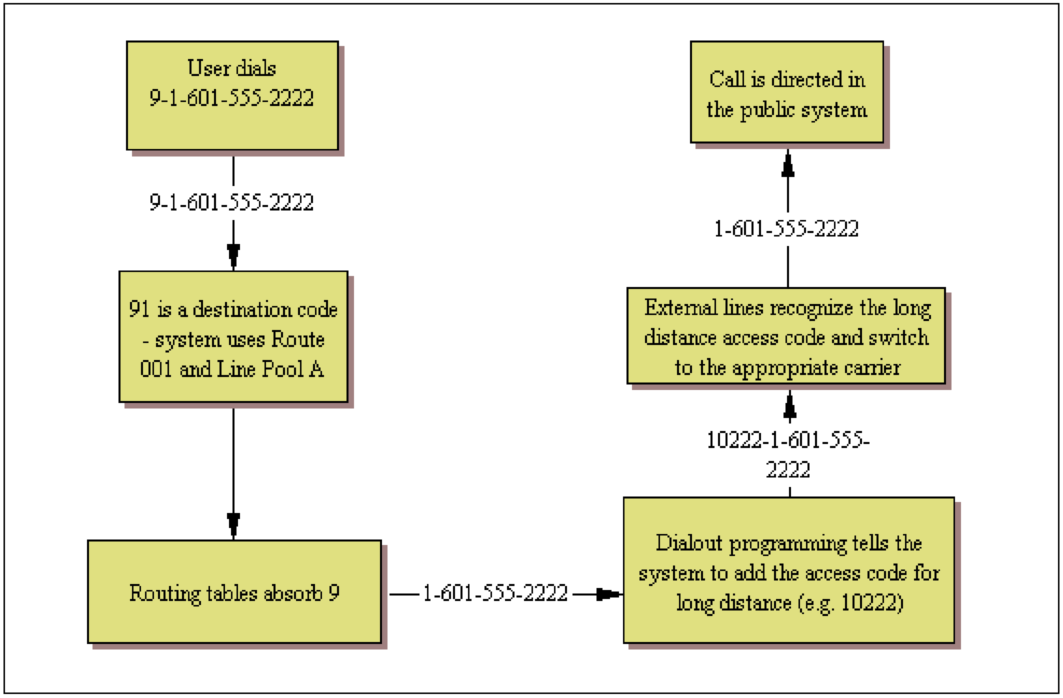
- **DN type:** displays for PRI lines with protocol set to SL-1 (MCDN, ETSI Euro).
- **Service type:** displays for PRI lines with protocol set to NI, DMS100, DMS250, 4ESS.
- **Service ID: N/A** appears where the service requires an ID.

Adding Carrier access codes to destination codes

In some cases, long distance service uses the same lines as local service but is switched to a specific carrier using an access number, which is sometimes referred to as an carrier access code (CAC). Route programming can include the access number so the users do not have to dial it every time they make a long distance call. Figure 90 shows an example of how the system interprets what the user dials into a valid outgoing call.

Note: Carrier code service must be supported from the Central Office.

Figure 90 Carrier code call numbering sequence


User dials 9-1-601-555-2222
9-1-601-555-2222
91 is a destination code - system uses Route 001 and Line Pool A
Routing tables absorb 9
1-601-555-2222
Dialout programming tells the system to add the access code for long distance (e.g. 10222)
10222-1-601-555-2222
External lines recognize the long distance access code and switch to the appropriate carrier
1-601-555-2222
Call is directed in the public system

To program a long distance carrier access code into a destination code

1. Create a route that uses a line pool containing local lines only. ("Routes" on page 290)
2. Program a route to use a line pool containing the lines used to access the long distance carriers.
3. Type the dialout digits, which are the same as the access digits. For example, if the access code is 10222, the dialout digits are 10222.
4. Create a destination code 91: 9 (for outside access) and 1 (for long distance). You must use a valid destination code.

5. Set **Absorbed Length** to 1.
   The digit 9 is only used internally and should be dropped. The 1 is needed to direct the call to the public carrier network.

Tips: The destination codes 9 and 91 used in the examples cannot be used together. If you need the destination code 91 to direct long distance calls, you must create a separate set of codes that use local calling routes. These codes would be, for example, 90, 92, 93, 94, 95, 96, 97, 98 and 99. Refer to "Grouping destination codes using a wild card" on page 281 for information on programming destination codes.

Using the MCDN access codes to tandem calls

Three special access codes exist specifically for programming calls over PRI and VoIP trunks that are using the MCDN protocol, and which connect to a call servers that use specific call codes for special call types, such as the Meridian 1 (M1). The purpose of the codes is to allow easier programming of the call servers when calls are tandemed through a BCM system to the local PSTN. Refer to "Private Network Settings" on page 318 for a description of these fields in context with the private dialing plan.

This is how the codes relate:

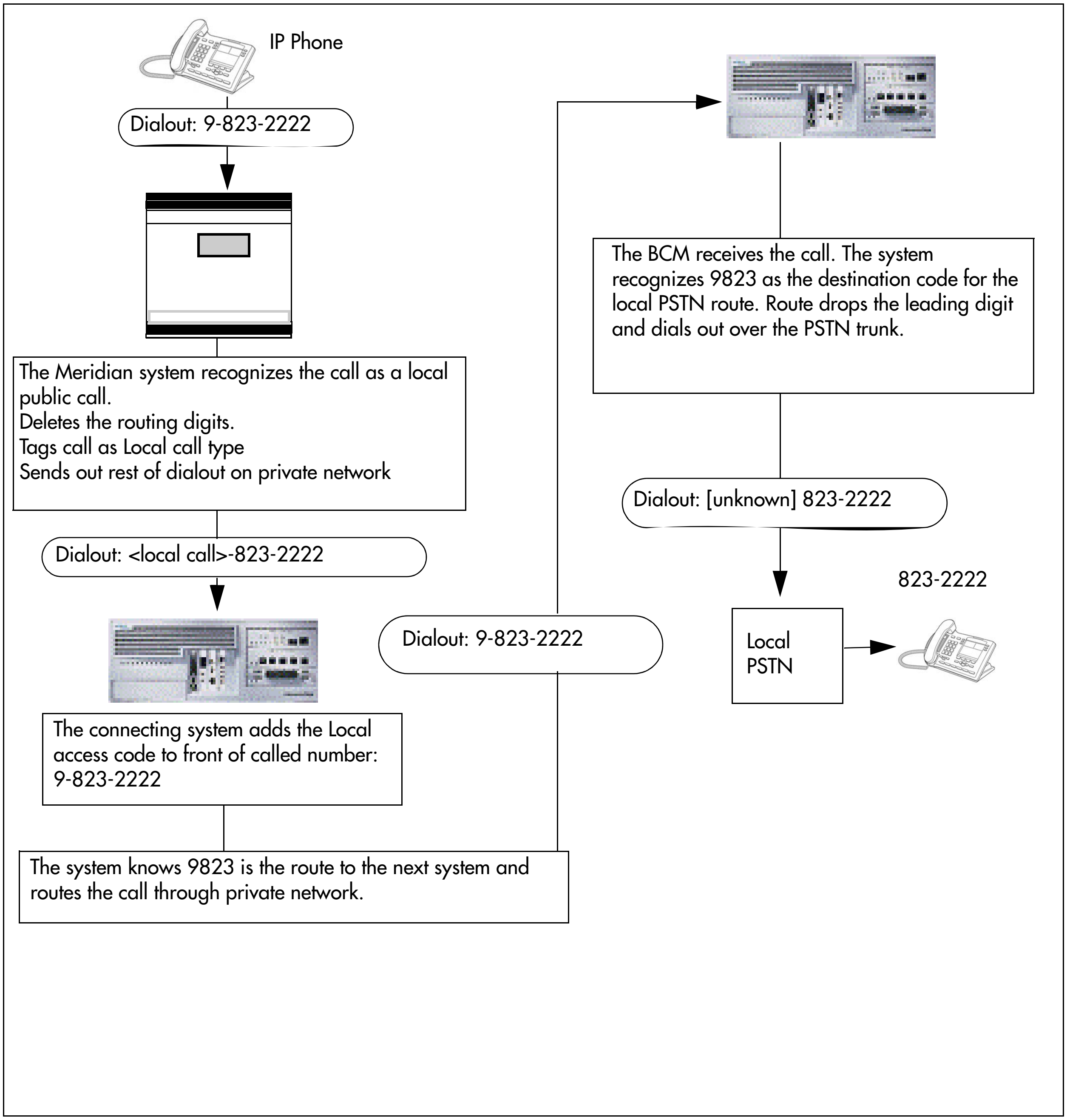
Meridian 1 access codes	BCM access codes	Sample code
Network/long distance code	Private access code	6
	National access code	61
Local code	Local access code	9
	Special access code	9

Calls tandeming to the public network through the private network need to retain their dialing protocol throughout network. This means that a call from an M1 node tagged as a local call gets received by the local node and is recognized as a call intended for the public network, but also as a call that needs to maintain the local tag until it gets to the local node that is directly connected to the PSTN. This is accomplished by ensuring that the destination code, which starts with this access code, passes the call on using the route designated with the correct DN type. Refer to "Setting up a route through a dedicated trunk" on page 280.

Calls coming in from the public network need to be translated to their private network destination before routing/tandeming through the private network. In this case, the route used is defined with the DN type of Private.

Figure 91 charts the process for a call tandeming through a BCM to the local public network.

Figure 91 Local call tandemed through private network nodes



The Meridian system recognizes the call as a local public call.
Deletes the routing digits.
Tags call as Local call type
Sends out rest of dialout on private network

The connecting system adds the Local access code to front of called number: 9-823-2222

The system knows 9823 is the route to the next system and routes the call through private network.

The BCM receives the call. The system recognizes 9823 as the destination code for the local PSTN route. Route drops the leading digit and dials out over the PSTN trunk.

Chapter 26

Dialing plan: Routing and destination codes

A large system usually requires a number of destination codes to ensure that calls are directed to the correct trunks, either on the private or public network.

The following paths indicate where to access destination codes in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > Routing**
- Telset interface: ****CONFIG > Services > Routing Services > Routes**

The following panels allow you to:

- create routes
- create destination codes for the routes, and the Normal schedule
- create alternate routing schedules

Click one of the following links to connect with the type of information you want to view:

Panels	Tasks	Feature notes
“Routes” on page 290	“Grouping destination codes using a wild card” on page 281	
“Destination codes” on page 292	“Using the MCDN access codes to tandem calls” on page 287	
	“Programming the PRI routing table” on page 285	
	“Setting up a destination for local calling” on page 279	
	“Setting up a route through a dedicated trunk” on page 280	
	“Adding Carrier access codes to destination codes” on page 286	
“Alternate routes for routing schedules” on page 294	“Programming for least-cost routing” on page 282	
	“Using multiple routes and overflow routing” on page 282	

Panels	Tasks	Feature notes

See also:

- “Setting up VoIP trunks for fallback” on page 423
- “Configuring lines” on page 147
- “BRI ISDN: BRI T-loops” on page 217
- “Dialing plan: System settings” on page 303
- “Dialing plan: Public network” on page 311
- “Dialing plan: Private network settings” on page 317
- “Dialing plan: Line pools and line pool codes” on page 297
- “Public networking: Tandem calls from private node” on page 327
- “Private networking: Using destination codes” on page 373
- “Private networking: PRI and VoIP tandem networks” on page 357
- “Private networking: MCDN over PRI and VoIP” on page 329
- “Private networking: DPNSS network services (UK only)” on page 365
- “Configuring centralized voice mail” on page 385

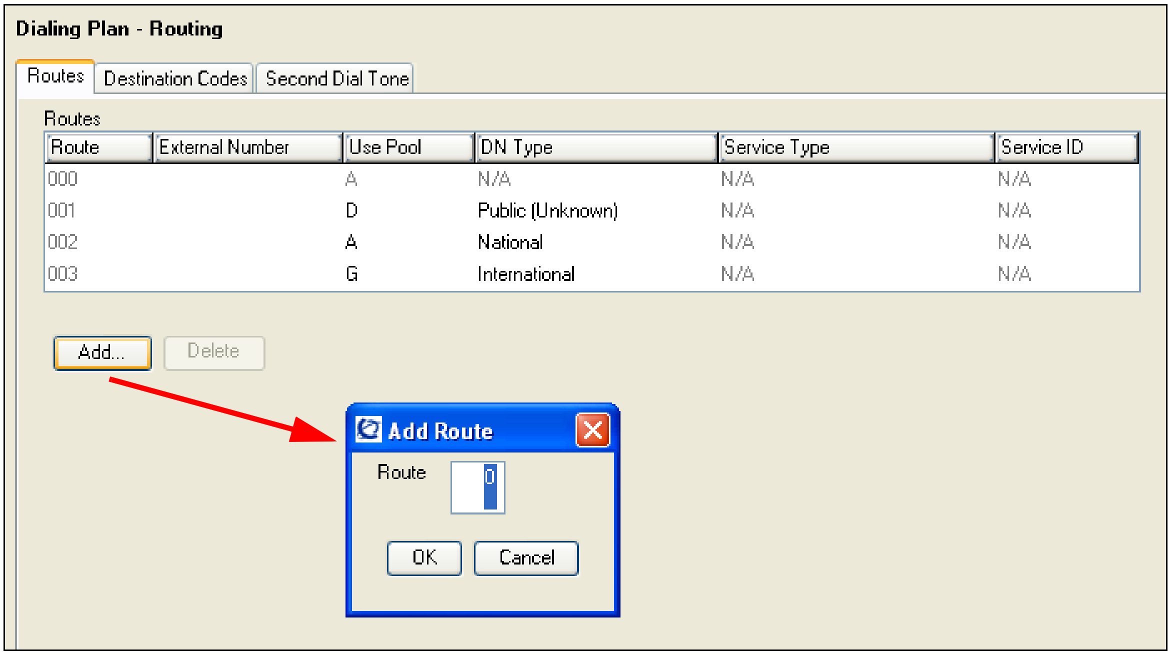
Click the navigation tree heading to access general information about DN records.

Routes

The first step to setting up call routing is to define line pools into uniquely named routes. A route can be used with more than one destination code, but a line pool should only be used with one route.

Figure 92 illustrates the Routes tab.

Figure 92 Routes tab



Dialing Plan - Routing

Routes

Route	External Number	Use Pool	DN Type	Service Type	Service ID
000		A	N/A	N/A	N/A
001		D	Public (Unknown)	N/A	N/A
002		A	National	N/A	N/A
003		G	International	N/A	N/A

Add Route

Route: 0

Table 63 describes the fields on the top frame.

Table 63 Route settings (Sheet 1 of 2)

Attribute	**Value**	**Description**
Route	<001-999>	This number is unique to each route.
External Number	<a maximum of 24 digits>	Enter the external or dial-out number for the route you want the assigned telephone to use. The external number is a digit or group of digits that get inserted in front of your dialed digits. If all the required numbers are defined in the destination code/dial string, this box can be left empty. Optional dial string entries: P = 1.5 second pause (counts as one digit in the dialing string) (F78 telset) DT = wait for dial tone (counts as two digits in the dialing string) (F804 telset)
Use Pool	Pool A to Pool O or BlocA to BlocF	Select a line pool for the route. The Bloc pools only display if you have PRI or ETSI QSIG trunks.
DN Type	Public Private Local (Subscriber) National Special (International)	This setting tells the system what type of line protocol the route uses to process the dial string. Refer to "PRI route types" on page 292. **MCDN private networks:** Local, National and Special are special designators used to route calls from Meridian 1 systems, through BCM systems, out to the public network. The codes for these settings are defined under Telephony > Dialing plan > General > Private Networks tab. Also refer to "Using the MCDN access codes to tandem calls" on page 287. When the BCM receives outgoing calls from the Meridian 1, it recognizes the call type and appends the appropriate access code to the Meridian dial string. This code then matches to a route that uses the same DN type, passing the call along, either to another node (the route would have the same DN type) or to the public network (the route would have a Public DN type), depending on the routing information.
Service Type	Public Private TIE Foreign exchange (FX) OUTWATS Switched Digital (SDS)	This setting tells the system what type of line protocol the route uses to process the dial string. These protocols are used for lines connected to DMS100, DMS250 and 4ESS switches. Refer to "PRI route types" on page 292.
Service ID	<digits>	If you choose a service, type in the identification number for the service.
Note:	**Outgoing call display:** If you have the trunks set up to send called number information, and the DN type is set to anything, except Private, the system sends the Public OLI number you specified under line programming. If the DN type is set to Private, the system sends the Private OLI number. Refer to "Line Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).	
Actions:		
Add	1. Under the routes table, click **Add**. 2. Enter a route number in the dialog box. 3. Click **OK** to save the new route.	
Delete	1. On the routes table, select the route you want to delete. 2. In the Routes panel, click **Delete.** 3. Click **OK**.	

Table 63 Route settings (Sheet 2 of 2)

Attribute	Value	Description
Modifying routes:	**Warning:** Modifying some route settings may result in dropped calls. Ensure that you modify the destination codes Absorbed Length setting, if required, if you add or change the External Number entry. Changing the Use Pool or DN Types/Service Types values will result in dropped calls if the lines in the line pool do not support the DN/Service Type selected. 1. On the routes table, select the route you want to change. 2. Click the field you want to change for that route and enter the new value. 3. Press Tab on your keyboard to save the change.	

PRI route types

Table 64 lists the service/DN type choices available for PRI lines.

Table 64 PRI Service type/DN type values

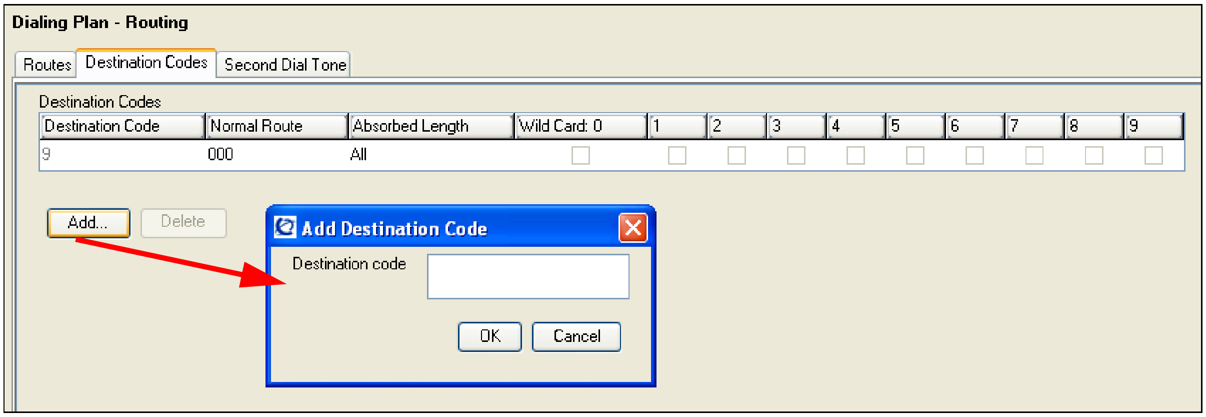
PRI Protocol	Type	Values
SL-1	DN	Public, Private, Local, National, Special
ETSI Euro	DN	None, Overlap
ETSI QSIG	N/A	
NI	Service	Public, TIE, Foreign Exchange (FX), OUTWATS
DMS100	Service	Public, Private, TIE, Foreign Exchange (FX), OUTWATS
DMS250	Service	Public, Private, TIE, Foreign Exchange (FX), OUTWATS
4ESS	Service	TIE, OUTWATS, Switched Digital (SDS)

Destination codes

Once you have the routes configured, set up the dialing plan destination codes that allow users to access the routes. You can use a route for more than one destination code, as you may require different codes for the same route to define restrictions or special call designators. Also see "Alternate routes for routing schedules" on page 294.

Figure 93 illustrates the Destination codes panel.

Figure 93 Destination codes table panel



Destination Codes

Destination Code	Normal Route	Absorbed Length	Wild Card: 0	1	2	3	4	5	6	7	8	9
9	000	All	□	□	□	□	□	□	□	□	□	□

Destination code

Table 65 describes the fields on the destination codes frame.

Table 65 Destination codes table

Attribute	Value	Description
Destination Code	<max. 12 digits>	This number precedes a telephone number to tell the system where the call needs to be routed. An *A* in the destination code represents an *any* character designation. The *A* code is a wildcard.
Normal Route	<configured route #>	This is the route that the system will use when the destination code is added to the dial string.
Absorbed Length	All, None, 1-X	This indicates how much of the destination code gets removed before the system sends the dial string to the network.
Wild Card 0 - 9	Included, Excluded, Unavailable	If you enter the wild card character *A* at the end of a destination code, then the following applies: Included: This number can be dialed as part of the destination code. Excluded: This number will not be accepted as part of a destination code string because it is already used in the system. Unavailable: This number is already defined in another destination code and cannot be used.
Actions		
Add	1. Under the Destination Codes table, click **Add**. 2. Enter the new destination code. 3. Click **OK** to save the route settings. 4. On the Destination Codes table, select the fields beside the route you just created, and modify them, as required. 5. Test the route.	
Delete	1. On the Destination Codes table, select the destination code you want to delete. 2. In the Destination Codes pane, click **Delete**. 3. Click **OK**.	

Note: The destination codes must not conflict with the following:

- park prefix
- external code
- direct dial digit
- Auto DN
- DISA DN
- Private access code
- line pool codes
- telephone DN
- public target line received digits
- other routing codes

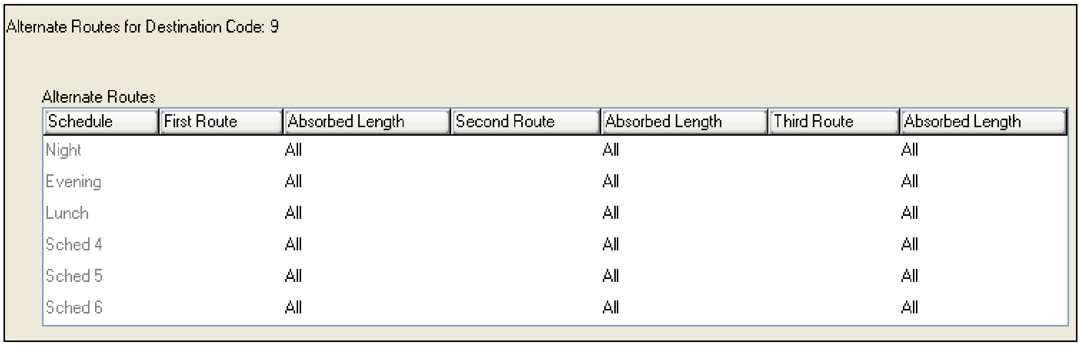
Alternate routes for routing schedules

When you select a route on the Destination Codes panel, the alternate schedules for that route appear in a separate table. You only need to fill out this panel if your system is using routing schedules.

Note that in these schedules you can configure three routes. The second route acts as fallback route for the first route if it is unavailable. If the second route is also unavailable, the system will try the third route. The dialing sequence for these routes needs to be the same from the user perspective, as fallback occurs automatically and is not controlled by the user. If all three routes fail, the default normal route is used.

Figure 94 illustrates the Alternate Routes panel.

Figure 94 Alternate routing schedules



Alternate Routes for Destination Code: 9

Alternate Routes

Schedule	First Route	Absorbed Length	Second Route	Absorbed Length	Third Route	Absorbed Length
Night		All		All		All
Evening		All		All		All
Lunch		All		All		All
Sched 4		All		All		All
Sched 5		All		All		All
Sched 6		All		All		All

Table 66 describes the fields on the Destination codes frame.

Table 66 Destination codes schedules (Sheet 1 of 2)

Attribute	Value	Description
Schedule	Defaults: Night, Evening, Lunch, Weekend, Sched. 5, Sched. 6	If you use a different carrier at different times of the day or week, you can set the destination code to use that route and provide two more backup routes. The user does not experience any difference in dialing sequence.

Table 66 Destination codes schedules (Sheet 2 of 2)

Attribute	Value	Description
First Route	<configured route #>	This is the route that the system will use, during the indicated schedule, when the destination code is added to the dial string.
Absorbed Length	All, None, 1-X	This indicates how much of the destination code gets removed before the system sends the dial string to the network.
Second Route	<configured route #>	This is the route the system will use if the first route is unavailable.
Absorbed Length	All, None, 1-X	This indicates how much of the destination code gets removed before the system sends the dial string to the network.
Third Route	<configured route #>	This is the route the system will use if the first and second route are unavailable.
Absorbed Length	All, None, 1-X	This indicates how much of the destination code gets removed before the system sends the dial string to the network.

Second Dial Tone

This feature provides dial tone for outgoing calls on any PRI line, based on the digits dialed. Digits dialed must match an entry in the second dial tone table to enable a second dial tone. Dial tone occurs on the line until another digit is dialed, a timeout occurs, or the user hangs up.

Up to 10 separate entries can be stored in the second dial tone table. The maximum digit length for each entry is four. Each entry must be unique and cannot conflict with:

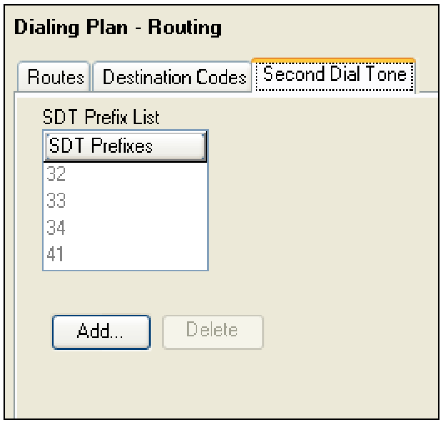
- Internal DNs
- Hunt Group DNs
- DISA DNs
- Auto DNs
- Target Line DNs

Tips: Entries can match destination or access codes for outgoing lines.

The following paths indicate where to configure the Second Dial Tone in Element Manager and through Telset Administration:

- Element Manager: **Telephony > Dialing Plan > Routing > Second Dial Tone**
- Telset interface: ****CONFIG > Services > Routing Service > 2nd Dial Tone**

Figure 95 Second Dial Tone



Dialing Plan - Routing

SDT Prefix List

SDT Prefixes
32
33
34
41

Table 4 Second Dial Tone

Attribute	Value	Description
SDT Prefix List		
SDT Prefixes		Enter the digits to match to trigger a second dial tone.
Actions		
Add	Button	Select to add an SDT prefix.
Delete	Button	Select an SDT prefix from the list and click delete to remove from the list.

➡ **Note:** Second dial tone is not provided on outgoing lines for remote access users and for ISDN terminal users when the Call Transfer feature is activated.

Chapter 27

Dialing plan: Line pools and line pool codes

The Line Pools panels allow you to:

- assign access codes to line pools
- add lines to line pools
- assign lines pools to telephones (and view which telephones have line pool assigned)
- set Call by Call limits for PRI service types

The following paths indicate where to access line pools settings in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > Line Pools**
- Telset interface: ****CONFIG > System Prgrming > Access Codes > Line pool codes**

Click one of the following links to connect with the type of information you want to view:

Panels	Tasks	Features and notes
"Line pools (and access codes)" on page 297		"Line pool access code notes:" on page 298
"Line pools: DNs tab" on page 299		
"Line pools: Call-by-Call Limits tab (PRI only)" on page 300		

Also refer to:

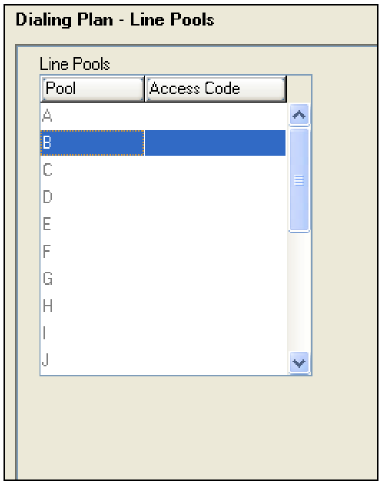
- "Configuring lines" on page 147
- "Dialing plan: Routing and destination codes" on page 289
- "Private networking: Using shared line pools" on page 347
- "Line Access - Line Pool Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600)

Click the navigation tree heading to access general information about DN records.

Line pools (and access codes)

The panel in the top frame displays settings that are configured on other panels. The only setting you can modify on this table is the access code number. Figure 96 illustrates this panel.

Figure 96 Dialing Plan - Line Pools table



Dialing Plan - Line Pools

Line Pools

Pool	Access Code
A	
B	
C	
D	
E	
F	
G	
H	
I	
J	

Table 67 describes the fields on the top frame.

Table 67 Line Pools table fields

Attribute	Value	Description
Pool	Read-only	These are the available line pools. Program only the ones for which you have actually assigned lines. Line pools are configured on the Lines panel
Access Code	<XXX>	Use access codes if you are not using destination codes on the system. These codes serve the same purpose, without the ability to define dialing sequences and multiple codes per route.

Line pool access code notes:

➡ **Note:** You cannot assign Bloc line pools with a line pool access code. You must define Bloc line pools under routing, and create destination codes for the routes.

➡ **Note:** A line pool access code cannot conflict with the following table.

Note: The line pool number must not conflict with the following:

- park prefix
- external code
- direct dial digits
- private access code
- Public/Private Auto DN
- Public/Private DISA DN
- Telephone DN

If the line pool code and the external code start with the same digit, the line pool code programming supersedes the external code.

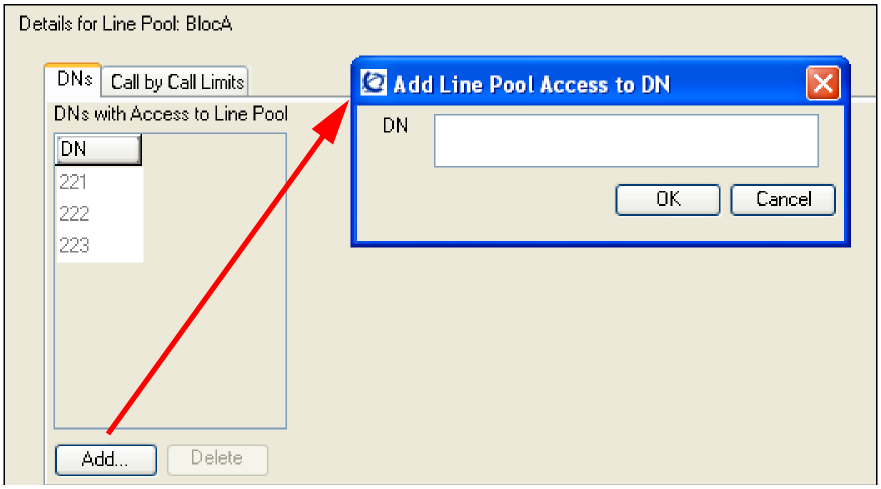
Line pools: DNs tab

The DNs tab shows you which DNs have this line pool assigned.

Programming note: A line pool must be assigned to a telephone before the user can use the line pool access code (or destination code) to make a call.

Figure 97 illustrates the DNs tab.

Figure 97 DN access to line pools



Details for Line Pool: BlocA

DNs with Access to Line Pool

DN
221
222
223

Table 68 describes the fields on the DNs tab.

Table 68 Line Pools: DN access to line pools fields (Sheet 1 of 2)

Attribute	Value	Description
DNs	Read-only	The telephones assigned to the line pool. Also refer to: Line Access - Line Pool Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).
Actions:		

Table 68 Line Pools: DN access to line pools fields (Sheet 2 of 2)

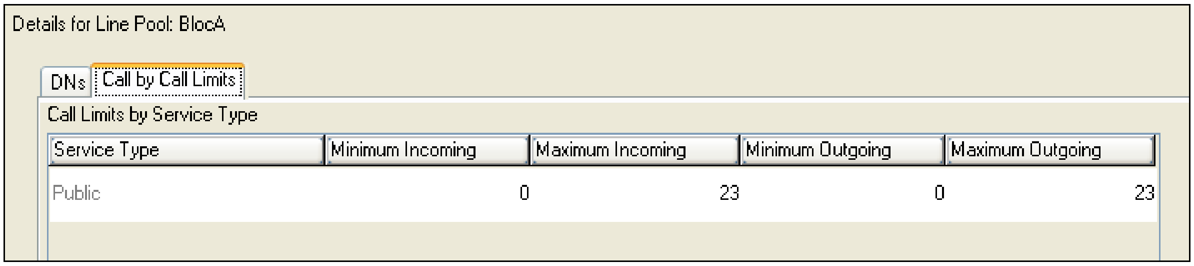
Attribute	Value	Description
Add	1. On the Line Pools table, select the line pool you want to modify. 2. Under the DNs tab table, click **Add**. 3. Enter the DN you want to assign to the line pool. 4. Click **OK** to save.	
Delete	1. On the Line Pools table, select the line pool you want to modify. 2. On the DNs tab table, select the DN you want to delete. 3. Under the DNs tab table, click **Delete**. 4. Click **OK**.	

Line pools: Call-by-Call Limits tab (PRI only)

For PRI lines that provide Call-by-Call services, Bloc line pools have an additional configuration that allows you to configure service type limitations. For information on PRI protocols, refer to Table 58.

Figure 98 illustrates the Call by Call Limits tab.

Figure 98 Line Pools: Call by call Limits fields



Details for Line Pool: BlocA

Call Limits by Service Type

Service Type	Minimum Incoming	Maximum Incoming	Minimum Outgoing	Maximum Outgoing
Public	0	23	0	23

Table 69 describes the fields on the Lines tab.

Table 69 Line Pools: Call by call limits fields

Attribute	Value	Description
Service Type	<read-only>	This is the type of CbC service provided on the PRI trunks in the line pool.
Minimum Incoming	Default: 2	**Note**: The total of the minimum values for incoming or outgoing PRI services cannot exceed the total number of lines in the Blocpool. The maximum value for an incoming or outgoing PRI service cannot exceed the total number of lines in the Bloc pool.
Maximum Incoming	Default: 23	
Minimum Outgoing	Default: 4	
Maximum Outgoing	Default: 23	

Chapter 28

Dialing plan: System settings

The panels described in this section define various common system settings that affect, or that are affected by, number planning.

The following paths indicate where to access system settings for dialing plans in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > General**
- Telset interface: ****CONFIG > System Programming > Access codes; System Programming > General > Direct Dial sets**

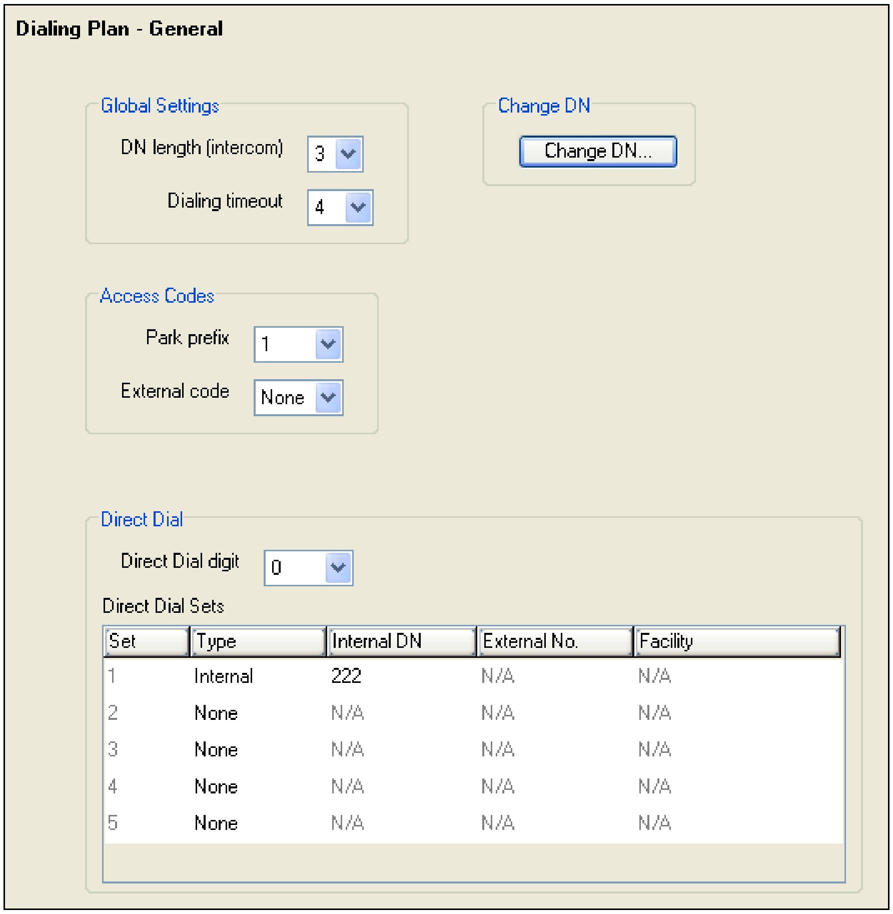
Panels/Subpanels	Tasks	Feature notes
"Common dialing plan settings" on page 303 • DN length • Dialing Time out • Park code • External code • Direct dial	"To define a direct dial set" on page 306 "Capabilities and Preferences - Capabilities tab" in the *BCM 4.0 Device Configuration Guide* (N0060600) (assign direct dial set to a telephone)	"Configuring CLID on your system" on page 235 "DN length constraints" on page 306 "Received number notes" on page 307 "Tips about access codes" on page 308 "Call Park codes" on page 309
Also refer to: • "Dialing plan: Public network" on page 311 • "Dialing plan: Private network settings" on page 317 • "Dialing plan: Line pools and line pool codes" on page 297		
Click the navigation tree heading to access general information about dialing plans.		

Common dialing plan settings

The fields on the Dialing Plan - General panel allow you to set some general system dialing features.

Figure 99 illustrates the Dialing Plan - General panel.

Figure 99 Dialing Plan - General settings and Direct Dial devices



Dialing Plan - General

Global Settings

DN length (intercom): 3

Dialing timeout: 4

Change DN

Access Codes

Park prefix: 1

External code: None

Direct Dial

Direct Dial digit: 0

Direct Dial Sets

Set	Type	Internal DN	External No.	Facility
1	Internal	222	N/A	N/A
2	None	N/A	N/A	N/A
3	None	N/A	N/A	N/A
4	None	N/A	N/A	N/A
5	None	N/A	N/A	N/A

Table 70 describes each field on this panel.

Table 70 Private and Public received numbers (Sheet 1 of 3)

Attribute	Value	Description
Global Settings		
DN length (intercom)	(3 to 7)	This is the length of the locally-dialed telephones. This field is set when the system is first configured. **Warning:** If this system is part of a private network, ensure that this value is compatible with the network requirements. This value is mirrored in the Private Received Number Length field for target lines. Refer to "Configuring lines: Target lines" on page 177. **Note:** If the DN length is changed, it will cause VM/CC to be defaulted in order to work properly.

Table 70 Private and Public received numbers (Sheet 2 of 3)

Attribute	Value	Description
Dialing timeout	Default: 4 seconds	This is the maximum period allowed between user dialpad presses before the system decides that the dial string is complete.
Access Codes		
Park prefix	None <one-digit number>	The Park prefix is the first digit of the call park retrieval code that a user enters to retrieve a parked call. If the Park prefix is set to None, calls cannot be parked. Refer to "Call Park codes" on page 309 before choosing a number. **SWCA note:** If this field is set to **None**, the system-wide call appearance (SWCA) feature will not work. Refer to "System Wide Call Appearances" in the *BCM 4.0 Device Configuration Guide* (N0060600).
External code	None <one-digit number>	The External code setting allows you to assign the external line access code for 7100 and 7000 digital phones and analog telephones attached to ATA 2s or to analog modules to access external lines. **Note:** Model 7000 phones are supported in Europe only. When the caller picks up the handset, the system tone sounds. The caller then enters this number to access an external line. **Note:** This number is overridden by line pool or starting with the same digit(s). Refer to "Tips about access codes" on page 308 before choosing a number.
Change DN		
Change DN	<button>	Click to reidentify a DN. **Note:** This method is faster than reidentifying the DNs under **Configuration > Telephony > Dialing Plan > DNs.**
Direct Dial		
Direct Dial digit	None <one-digit number>	The Direct dial digit setting allows you to specify a single system-wide digit to call a direct dial telephone.
Define Direct Dial Sets: Refer to "To define a direct dial set" on page 306.		
Set	<1-5>	This tags the telephone to the system.
Type	Internal External None	This is the type of number for the direct-dial set.
Internal DN	DN	The DN number of the telephone to be designated as the direct dial set. (Internal sets).
External No.	<external dial string>	The actual phone number, including destination codes, of the direct dial set (External sets).

Table 70 Private and Public received numbers (Sheet 3 of 3)

Attribute	Value	Description
Facility	Line Pool (A-O) Use prime line Use routing table	The facility to be used to route the call to a direct dial set that you define with an external number. **Note:** If you choose **Use prime line**, ensure that prime line is not assigned to the intercom buttons for your telephones. When prime line is assigned as an intercom button, it chooses the first available line pool assigned to the telephone to make a call. If this line pool does not have the correct lines for routing the call, the direct dial call will fail. Refer to "Line Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

To define a direct dial set

1. On the Direct Dial table, click the fields beside the set number you want to configure and enter the appropriate values.
2. Press **Tab** to save the values.
3. Go to the DN records of the telephones where you want the direct dial set assigned and assign the set under "Capabilities and Preferences - Preferences tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

➡ **Note:** The BCM cannot verify that the number you assign as an external direct dial set is valid. Check the number before assigning it as a direct dial set by calling the direct dial you have assigned.

Configuration notes and tips

The information in this section expands on some of the fields found on the tabs on the Dialing Plan - General panel.

- "DN length constraints" on page 306
- "Received number notes" on page 307
- "Tips about access codes" on page 308
- "Call Park codes" on page 309

DN length constraints

⚠ **Warning:** Do not change DN length immediately after a system startup. You must wait until the system is operational with two solid green status LEDs.

⚠ **Warning:** Increasing the DN length affects other areas of the system:

If the DN length change creates a conflict with the Park prefix, external line access code, direct-dial digit, or any line pool access code, the setting for the prefix or code changes to None, and the corresponding feature is disabled.

Optional applications affected by DN length changes:

Voice mail and **Contact Center** applications are reset if you change the DN length after these services are installed.

⚠ **Warning:** If your system is running with a PBX telephony template, the Public and Private received number length are set to 3 (digits) at start-up. Increasing the DN length after system startup does not change these digits, so you will need to manually change the Public and Private received number length.

Private OLIs are automatically assigned to the DN records if the DN length and the Private received number length are the same. If this changes, the Private OLIs are cleared, or are not assigned (PBX template).

Network note: If your system is part of a private network, ensure that you confirm the dialing plan for the network before changing this length. If you change the length, ensure that you check all DN-related settings after the change.

Received number notes

- If you change the received number length for your system, the **Public number** entry for the target lines will clear if the new received # length is less than the number entered in this field.
- If the new received number length has more digits than the number entered in the target lines Public Number field, the entry remains, but does not update to the new DN length.
- A private OLI is automatically assigned to the DNs if the DN length and the Received number length are the same. If either changes so that they are not the same, the private OLI field is cleared or not assigned (PBX template).

Tips about access codes

Here are some pointers to assist you in planning the access codes for your system.

Note: The following values must not conflict:

- Park prefix
- external code
- direct dial digit
- Private access code
- Public/Private Auto DN
- Public/Private DISA DN
- line pool code/destination code
- telephone DN

Note: If the line pool code and the external code start with the same digit, the line pool code programming supersedes the external code.

External line access code:

Example: If you enter the following selections:

Park Prefix - 1

Direct Dial digit - 0

Telephone DNs - 2000-2500

You wish to add a destination code of 2500 and 12. This cannot be accomplished as this would conflict with existing dialing numbers. To solve this you could modify the Park prefix and change the Telephone DN of 2500.

- If the DN length is changed, and the changed DNs conflict with the external line access code, the setting changes to None.
- **Direct dial telephone:** Another direct dial telephone, an extra dial telephone, can be assigned for each schedule in Services programming.

  If the DN length is changed, and the changed DNs conflict with the Direct dial digit, the setting changes to None.
- **Public/Private Auto DN:** The length of the Auto DNs are the same as the Public or Private Received Number Lengths specified under Telephony > Dialing Plan > General. The public/private Auto DN is cleared if the corresponding Received Number Length is changed.
- **Public/Private DISA DN:** The length of the DISA DNs are the same as the Public or Private Received number length specified under Telephony > Dialing Plan > General. The public/private DISA DN is cleared if the corresponding Received number length is changed.

Call Park codes

When you park a call (**FEATURE 74**), the system assigns one of 25 codes for the retrieval of the call. You can then press the `Page` display key to announce the code that appears on the display.

These three-digit codes include the Call Park prefix, which can be any digit from 1 to 9, and a two-digit call number between 01 and 25. For example, if the Call Park prefix is 1, the first parked call is assigned Call Park retrieval code 101.

Note: The park prefix must not conflict with the following:

- park prefix
- external code
- Direct dial digit
- Private access code
- Public/Private Auto DN
- Public/Private DISA DN
- line pool code/destination code
- telephone DN

Note: Other programmable settings may affect what numbers appear in the window during programming. Although the numbers 0 to 9 are valid Park prefix settings, some may already be assigned elsewhere by default or by programming changes.
If the DN length changes, and the changed DNs conflict with the Park prefix, the setting changes to None.

The system assigns Call Park codes to calls in sequence, from the lowest to the highest, until all the codes are used. A round-robin method means the use of different of codes ensures a call reaches the right person, especially when more than one incoming call is parked.

The highest call number (the Call Park prefix followed by 25) is used by model 7000 and 7100 telephones, analog telephones, or devices connected to the system using an ATA2. Analog telephones or devices cannot use the other Call Park codes.

Note: Model 7000 phones are supported in Europe only.

Calls are retrieved by pressing the intercom button and dialing the retrieval code. On model 7000 and analog telephones, pick up the receiver, and then dial ***<parkcode>*25**.

You also need to program the delay timer that determines when external parked calls that are not answered return to the originating telephone. Refer to “Timers” in the *BCM 4.0 Device Configuration Guide* (N0060600).

You can disable Call Park by setting the Park Code to None.

Chapter 29
Dialing plan: Public network

The panel described in this section defines the number planning required for calls exiting the system to the public telephone network.

The following paths indicate where to access the dialing plan for public network in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > Public Network**
- Telset interface: ****CONFIG > System Settings > Dialing Plan > Public**

Panels/Subpanels	Tasks	Feature notes
"Public dialing plan settings" on page 311		
"Public Network Settings" on page 312		
"Public network DN lengths" on page 313	"Adding a DN Prefix for public dialing" on page 314 "Modifying a DN prefix" on page 314 "Deleting a DN prefix" on page 315	"Outgoing public calls routing" on page 315
"Carrier Codes" on page 315	"To add a carrier code" on page 316 "To modify a carrier code" on page 316 "To delete a carrier code" on page 316	
"Configuration notes and tips" on page 306		

See also:

- "Dialing plan: System settings" on page 303
- "Dialing plan: Private network settings" on page 317
- "Public networking: Setting up basic systems" on page 323

Click the navigation tree heading to access general information about dialing plans.

Public dialing plan settings

The Dialing Plan - Public Network panel displays the fields that determine dialing information specific to dialing in or out to a public network from the host system.

This panel includes information about:

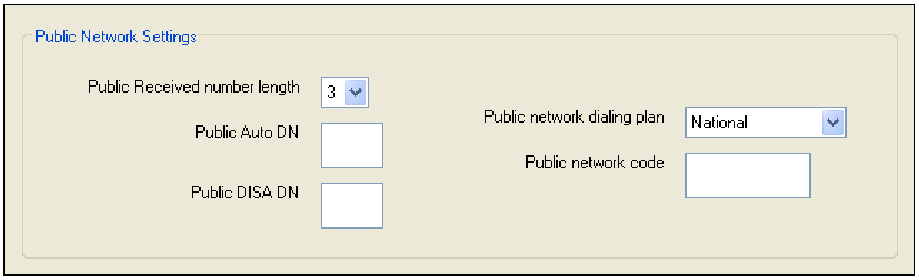
- “Public Network Settings” on page 312
- “Public network DN lengths” on page 313
- “Carrier Codes” on page 315

Public Network Settings

The following describes system settings that allow the system to determine if an incoming call is meant for the local system. These settings are used to determine how many digits the system needs to receive before sending the dial string over the trunk interface.

Figure 100 illustrates the Public Network Settings panel.

Figure 100 Public Network Settings panel



Public Network Settings

Public Received number length: 3

Public Auto DN:

Public DISA DN:

Public network dialing plan: National

Public network code:

Table 71 describes each field in the Public Network Settings box.

Table 71 Private and Public received numbers (Sheet 1 of 2)

Attribute	Value	Description
Public Received number length (max)	2, 3, 4, 5, 6, 7, 8, 9, 10, 11,12	The maximum number of digits (2, 3, 4, 5, 6, 7, 8, 9, 10, 11, 12) that the system uses to determine if an incoming call tagged as public fits the system public DN numbering. Default: DID template, same as DN length; PBX template: 3 Also refer to “Setting up a destination for local calling” on page 279.
Public Auto DN	<DN digits to be received from the auto-answer trunk>	Public network calls answered without DISA require no password to access the BCM. The type of service that applies to the call depends on the restrictions assigned to the trunk.
Public DISA DN	<DISA DN digits to be received from the auto-answer trunk>	For public network calls answered with DISA, the system presents a stuttered dial tone to prompt a caller to enter a valid password. The Class of Service (CoS) that applies to the call is determined by this CoS password. After a remote user accesses the BCM, they can change the existing CoS using the DISA DN. This gives you greater flexibility when you create access privileges. For example, you may want to have a shared DN for remote access, but separate CoS passwords with different dialing out privileges for individuals.
Public network dialing plan	National Local (subscriber)	Local dialing plan defines a seven-digit numbering scheme. National dialing plans define an extended number scheme. North America is set to 10 digits. However, systems in other countries may have a variable length.

Table 71 Private and Public received numbers (Sheet 2 of 2)

Attribute	**Value**	**Description**
Public network code	<1 to 8 digits>	This number concatonates with the Public OLI, which, by default, is the DN of the device. **Note**: In systems running the North American profile, if the Public OLI contains the public network code, that entry overrides any entry in this field. Refer to “Line Access tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).

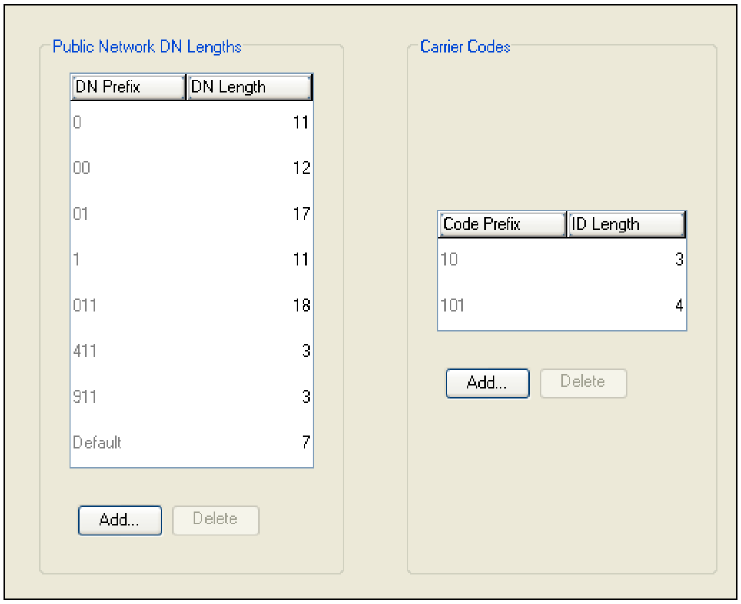
Public network DN lengths

The Public network DN length tells the system how long dialing strings will be when entering the network. For example, if you dial 18005551212 the public network DN length for 1, which is 11, tells the system to wait until 11 digits are entered before processing the call.

Note: If the values for Public Network DN length are set too short, digits will be stripped from the dialing string. Conversely, if the values are set too large, the dialing will take longer to process.

The Public Network DN Lengths/Carrier Codes panel allows you to define DN prefixes and define the length of the prefixes for public dialing. Figure 101 illustrates this panel.

Figure 101 Public Network DN Lengths/Carrier Codes panels



Public Network DN Lengths

DN Prefix	DN Length
0	11
00	12
01	17
1	11
011	18
411	3
911	3
Default	7

Carrier Codes

Code Prefix	ID Length
10	3
101	4

Table 72 describes each field on this panel.

Table 72 Public network DN values

Attribute	Values	Description
DN Prefix	<XXXX>	This is the number that must precede a dial string exiting the system to the public network. Each prefix defines a specific destination or type of call.
DN Length	<1-24>	This number indicates how many numbers, starting from the front of the dial string, the system will wait before sending to the public network.

About the Public Network DN lengths table

In the public Network DN lengths table:

- You can define up to 30 entries.
- Each entry consists of a DN prefix string (1 to 10 digits) and a length value (two digits, 1 - 25).
- Several entries are predefined in the North America profile. These defaults can handle most regions in North America without the need for additional programming. If required, you can remove or modify these entries.
- The table always contains one default entry. You cannot remove this entry. You can only modify the length parameter associated with this entry. The default entry specifies the length of any dialing string that does not match one of the other table entries.

Adding a DN Prefix for public dialing

The Default DN prefix cannot be deleted. The DN length for this prefix varies, depending on the country profile running on the system.

To add a new Prefix, follow these steps.

1. In the Public Network DN Lengths box, click **Add**.
2. Enter the new parameters:
   - DN Prefix
   - DN Length
3. Click **Save**.

Modifying a DN prefix

You can only change the DN length for a prefix. To change the prefix itself, delete the existing prefix and enter a new one.

1. On the Public Network DN Lengths panel, click the DN prefix you want to modify.
2. Click in the DN length field for that prefix and enter the new value.
3. Press Tab on your keyboard to save the setting.

Deleting a DN prefix

➡ **Note:** Dialing prefixes are used system-wide for users to make calls. Delete prefixes with caution.

1. On the Public Network DN Lengths panel, click the DN prefix you want to delete.
2. Click **Delete**.
3. Click **OK** on the confirmation dialog.

Outgoing public calls routing

Outgoing public calls from within the system typically have the routes set to Public. Refer to "Setting up a destination for local calling" on page 279. The NPI/TON gets sent as Unknown/ Unknown. The public called number length is based on the Public DN lengths table in the Public networks dialing plan.

MCDN trunks also allow public call types when tandeming calls from another system on the private network. Some of these systems use specific call types that the BCM needs to recognize to pass on correctly. Also refer to "Using the MCDN access codes to tandem calls" on page 287.

Type of call	NPI/TON	BCM prepend access code	BCM monitor display
Local	E164/Local	Local access code (9)	E.164/Subscriber
National	E164/National	National access code (X1)	E.164/National
Special calls (international, 911, etc.)	Private/Special	Special access code (9)	

Carrier Codes

The Carrier Codes table allows you to enter a maximum of five carrier code prefixes.

- You can define up to five carrier codes.
- Entries may be predefined for a specific country profile, but you can remove these defaults.
- Each entry consists of an equal access identifier code prefix (one to six digits) and a carrier identification code length (one digit, 1 to 9).
- Each entry is identified by the prefix digits themselves.

Table 73 describes each field on this panel.

Table 73 Carrier Code values

Attribute	Values	Description
Code Prefix	<one to six digits> (Read-only)	This value defines the prefix that will be used to access the carrier code.
ID Length	1, 2, 3, 4, 5, 6, 7, 8, or 9	This value defines the carrier ID length.

To add a carrier code

1. Click **Add**.
2. Enter the required code and ID:
   - Code Prefix
   - ID length
3. Click **Save**.

To modify a carrier code

1. Click the line for the Carrier Code where you want to change information.
2. Click the field that you want to change, and enter the new value.

To delete a carrier code

1. Click the line for the carrier code that you want to delete.
2. Click **Delete**.
3. Click **OK**.

Chapter 30
Dialing plan: Private network settings

The following panels define various system settings that affect or that are affected by number planning for private networks.

The following paths indicate where to access the dialing plan for private networks in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Dialing Plan > Private Network**
- Telset interface: ****CONFIG > System Prgrming > Dialing Plan > Public Network**

Panels/Subpanels	Tasks/Features
"Private Network dialing plan settings" on page 317	
"Private Network Settings" on page 318	"Outgoing private calls routing" on page 322
"Private Network - MCDN network (PRI SL-1, PRI ETSI, VoIP)" on page 319	
"ETSI-specific network features" on page 321	
"Configuration notes and tips" on page 306	
Also refer to:	
• "Dialing plan: System settings" on page 303 • "Dialing plan: Public network" on page 311 • "Private networking: Basic parameters" on page 349 • "Private networking: Using shared line pools" on page 347 • "Private networking: Using destination codes" on page 373 • "Private networking: PRI Call-by-Call services" on page 377 • "Private networking: PRI and VoIP tandem networks" on page 357 • "Private networking: MCDN over PRI and VoIP" on page 329 • "Private networking: MCDN and ETSI network features" on page 353 • "Private networking: DPNSS network services (UK only)" on page 365	
Click the navigation tree heading to access general information about dialing plans.	

Private Network dialing plan settings

The boxes on the Private Network Settings panel have fields that apply specifically to private network configurations. Network configurations can be set up between BCM4.0 systems and other call servers such as the Business Communications Manager, Meridian 1, or Succession 1000.

Some of the settings on this panel also depend on the market profile of the system.

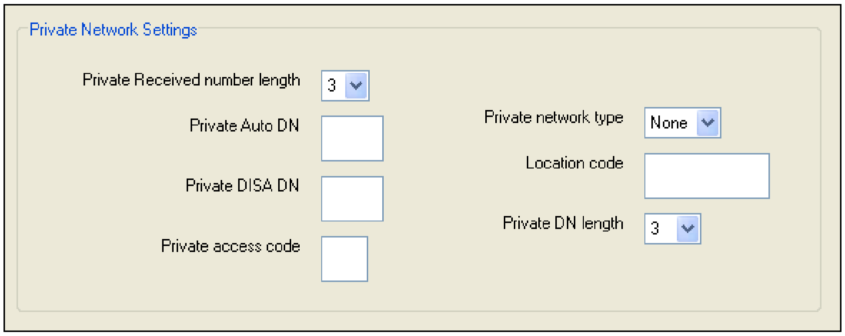
- "Private Network Settings" on page 318
- "Private Network - MCDN network (PRI SL-1, PRI ETSI, VoIP)" on page 319
- "VoIP-specific private network dialing" on page 320

- “ETSI-specific network features” on page 321

Private Network Settings

The settings on the Private Network Settings panel describe the numbering that the system uses to assess an incoming call to determine if the call is destined for your system or needs to be routed elsewhere on the private or public network. This panel is illustrated in Figure 102.

Figure 102 Private Network Settings panel



Private Network Settings

Private Received number length: 3

Private Auto DN:

Private DISA DN:

Private access code:

Private network type: None

Location code:

Private DN length: 3

Table 74 describes each field on this panel.

Table 74 Private Network Settings (Sheet 1 of 2)

Attribute	Value	Description
Private Network Settings		
Private Received number length	2, 3, 4, 5, 6, 7	The number of digits of an incoming dial string that the system uses to determine if a call tagged as Private fits the system private DN numbering. Default: DID template, same as DN length; PBX template: 3
* Private Auto DN	Digits to be received from a private auto-answer trunk>	Private network calls answered without DISA require no password to access the BCM. The type of service that applies to the call depends on the restrictions assigned to the trunk.
* Private DISA DN	<DISA DN digits to be received from the auto-answer trunk>	For private network calls answered with DISA, the system presents a stuttered dial tone to prompt a caller to enter a valid password. The Class of Service (CoS) that applies to the call is determined by this CoS password. After a remote user accesses the BCM, they can change the existing CoS password using the DISA DN. This gives you greater flexibility when you create access privileges. For example, you may want to have a shared DN for remote access, but separate CoS passwords with different dialing out privileges for individuals.
Private access code	<systemcode> **MCDN:** coordinate with National access code	This code identifies this system to the private network. It comes in as the first digit in a dial string defined as private and is read based on the private DN length. Example: if the dialed number is 7880, and the private DN length is 4, the system scans the four digits from the right, recognizing the 7 as the private access code for this system.

Table 74 Private Network Settings (Sheet 2 of 2)

Attribute	Value	Description
Private network type	CDP, UDP, None	Specify if your Private network uses a coordinated dialing plan (CDP) or a universal dialing plan (UDP). If you choose None, the private networking supplementary services are not available.
Private Network ID (CDP/UDP networks)	<1-127>	This is the unique number that identifies the system to the Meridian PRI-MCDN network. Both end points must match on a PRI-MCDN network. On a VoIP trunking-MCDN network, this ID must be the same on all nodes. This number is supplied by the private network administrator.
Location code	<unique three-digit number>	This code identifies this particular system for calls within the network for a UDP dialing plan. This number must be unique. **Note**: The system uses the Private Access Code length, plus the Location code length, plus the DN length to determine the DN length required to determine that a call is a private network call.
*Private DN length	<3-14>	The Private DN length parameter specifies the length of a dial string that the system uses to determine that the call is a private network call, when the route uses DN Type: Private.

* CDP and UDP private DN lengths are determined this way:

CDP: the system uses the telephone DN length

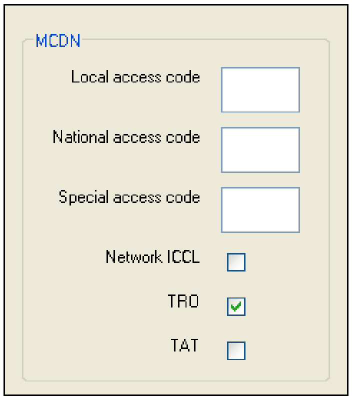
UDP: the system combines the private access code length + location code length + telephone DN length. When a call comes in, the system recognizes the leading digits as a private call and removes (truncates) them, leaving the telephone DN, which is recognized as the private DN length.

Private Network - MCDN network (PRI SL-1, PRI ETSI, VoIP)

If your system is part of a private network using the MCDN protocol, you may need to configure these special dialing access codes and network settings.

Figure 103 illustrates the MCDN panel.

Figure 103 MCDN network values



MCDN

Local access code

National access code

Special access code

Network ICCL ☐

TRO ☑

TAT ☐

Table 75 describes the values for these fields.

Table 75 Private network values

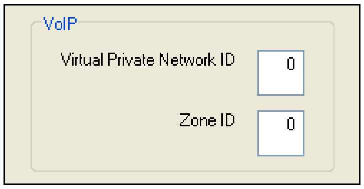
Attribute	**Values**	**Description**
Private networking also provides access to tandem calling and toll bypass functionality to users calling into the system. For example, a PSTN user in Toronto could call a PSTN user in Ottawa and have the call routed over the private network connection from the Toronto office to the Ottawa office and then out to the PSTN from the Ottawa office. This bypasses any long distance toll charges. BCM to BCM to PSTN: Calls are routed as private over the private network, and then flagged as public to go out to the end node PSTN. Meridian to BCM to PSTN: Special call codes from the Meridian (Local, National and Special access codes) need to be recognized by the BCM and correctly passed to the local PSTN.		
Local access code	<code to access local PSTN>	MCDN connections only. This number is prepended to an incoming M1 local dial string and designates the call as a Local call type (typically 9). Refer to “Using the MCDN access codes to tandem calls” on page 287.
National access code	<private access code + 1>	MCDN connections only. This number is prepended to an incoming call marked as a long distance call, and designates the call as a National type call (private access code + 1).
Special access code	<code to access local PSTN>	MCDN connections only. This number is prepended to an incoming international (011....) or special-case dial string (911, 411) and designates the call as a special type call (9011...., 9911, 9411).
Incoming and tandem calls (Also refer to “Dialing plan: Routing and destination codes” on page 289).		
Network ICCL	<check box>	ISDN Call Connection Limitation is part of the call initiation request. This feature acts as a check at transit PBX points to prevent misconfigured routes or calls with errors from blocking channels.
TRO	<check box>	Trunk Route Optimization occurs during the call setup. This feature finds the most direct route through the network to send a call between nodes.
TAT	<check box>	Trunk anti-tromboning works during an active call to find the optimum routing.
These features require compatible programming on the remote system.		

VoIP-specific private network dialing

The features contained in the VoIP subpanel are required for installations like the Survivable Remote Gateway (SRG), where the remote call server requires bandwidth management to handle calls.

Figure 104 illustrates the VoIP panel.

Figure 104 VoIP special dialing plan settings



VoIP

Virtual Private Network ID: 0

Zone ID: 0

Use Table 76 to determine the settings you want to define network services feature availability.

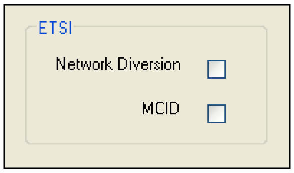
Table 76 VoIP special dialing plan values

Attribute	Values	Description
Virtual Private Network ID	<digits>	This is the VPN ID for a remote system, such as Succession 1000/M. In some applications, such as for the Survivable Remote Gateway (SRG) acting as a Branch Office, this ID is required to ensure that Bandwidth Management is handled correctly for calls coming into the Succession 1000/M from your system. See "Virtual Private Networks (VPN)" on page 641 for more information. Default: 0
Zone ID	<digits>	A remote system, such as Succession 1000/M, may configure your system into a separate zone to accommodate specific dialing requirements, such as for an SRG system acting as a Branch Office to a Succession 1000/M system. The system administrator of the Succession 1000/M system provides the Zone ID. Enter that number here and include it in any destination codes directed to, or through, that system so that the remote system can correctly direct incoming calls. Default:0

ETSI-specific network features

The features contained in the ETSI subpanel are service provider-based network services available for some PRI-ETSI lines. This subpanel is illustrated in Figure 105.

Figure 105 ETSI private network settings



ETSI

Network Diversion ☐

MCID ☐

Use Table 77 to determine the settings you want to define network services feature availability.

Table 77 ETSI private network settings fields (Sheet 1 of 2)

Attribute	Values	Description
Network Diversion	<check box>	Choose if you want to allow calls to be redirected to an outside network.

Table 77 ETSI private network settings fields (Sheet 2 of 2)

Attribute	Values	Description
MCID	<check box>	If you select this check box, the called party can use **FEATURE 897** to request the service provider network to record the identity of an incoming call. Including: • called party number • calling party number • local time and date of the activity • calling party sub-address, if provided by the calling user
		MCID note: The feature code must be entered within 25 seconds of the caller hanging up (a 25-second busy tone occurs). If the called party hangs up first, there is no opportunity to use the feature. **Note**: The call identification comes from your service provider, not the local system. You must have the service activated by the CO before the feature is active for the user, regardless of the setting in this field.

Outgoing private calls routing

When you set up routing for private calls, the route is set to Private. Refer to "Setting up a route through a dedicated trunk" on page 280.

How the system identifies the call depends on the type of trunk chosen for the route. Refer to the table below.

Dialing plan setting	NPI/TON	Private called number length based on
MCDN trunks send private calls in this way:		
None	Private/Subscriber	Private DN length (set on Private Network panel)
UDP	Private/UDP	private access code + home location code (LOC) + private received digits
CDP	Private/CDP	private received digit
DMS100/DMS250/ETSI-QSIG trunks send private calls in this way:		
None	Private/Subscriber	Private DN length (set on Private Network panel)
UDP	Private/Subscriber	private access code + home location code (LOC) + private received digits
CDP	Private/Subscriber	private received digit

Chapter 31

Public networking: Setting up basic systems

Public networks are the connection between the BCM and the public network (PSTN network).

This following provides examples of two basic types of systems.

- "Public networks: PBX system setup" on page 323
- "Public network: DID system" on page 324

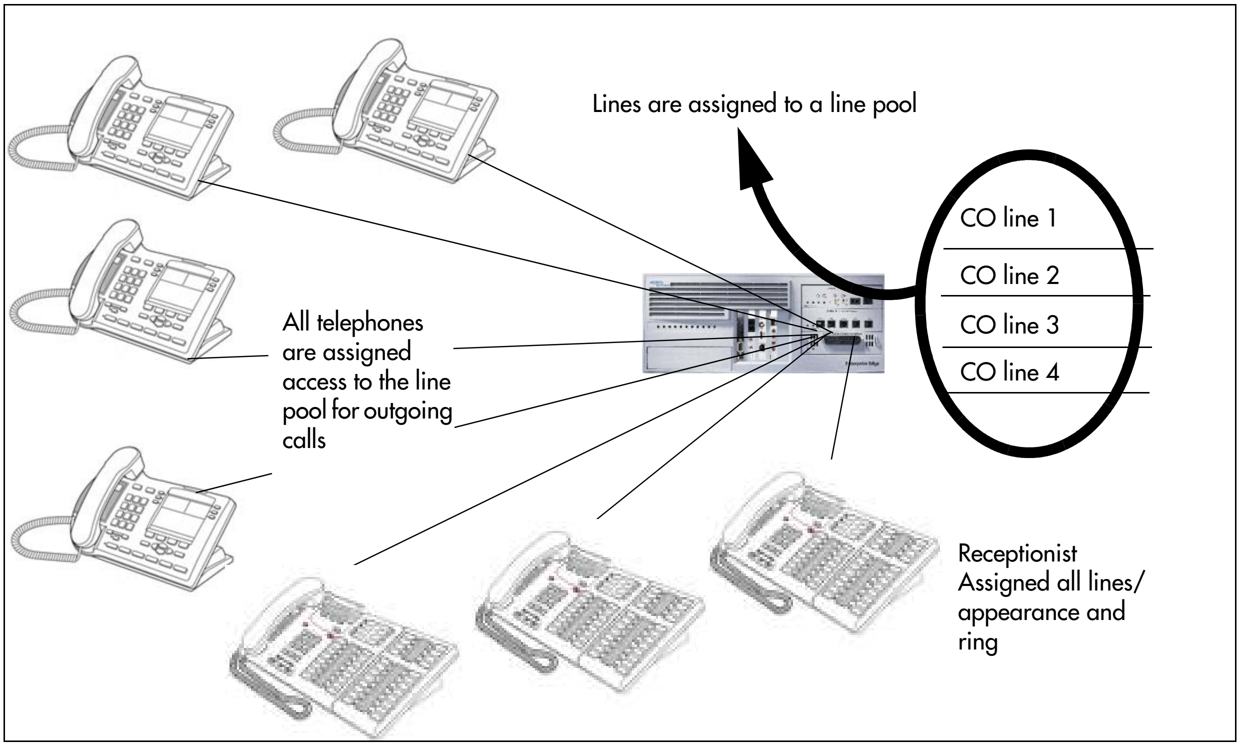
Public networks: PBX system setup

PBX is Short for Private Branch Exchange, a private telephone network used within an enterprise. Users of the PBX share a certain number of outside lines for making telephone calls external to the PBX. Dialing within the PBX is typically 3 to 4 digit dialing between local and remote networked nodes.

This setup is for a larger offices which have fewer CO lines than there are telephones. In this case the lines are pooled, and the line pool is assigned to all telephones. As well, there is a designated attendant with a telephone that has all lines individually assigned.

Figure 106 illustrates an example of a PBX system.

Figure 106 PBX system


Lines are assigned to a line pool
CO line 1
CO line 2
CO line 3
CO line 4
All telephones are assigned access to the line pool for outgoing calls
Receptionist Assigned all lines/ appearance and ring

Programming:

Lines

- Set lines to manual answer.
- Configure into a line pool.

Telephones

- Line pools are assigned to general office telephones.
- The Prime line is set to the line pool.
- Lines are assigned individually and as a line pool to the central answer position (set to appear and ring).

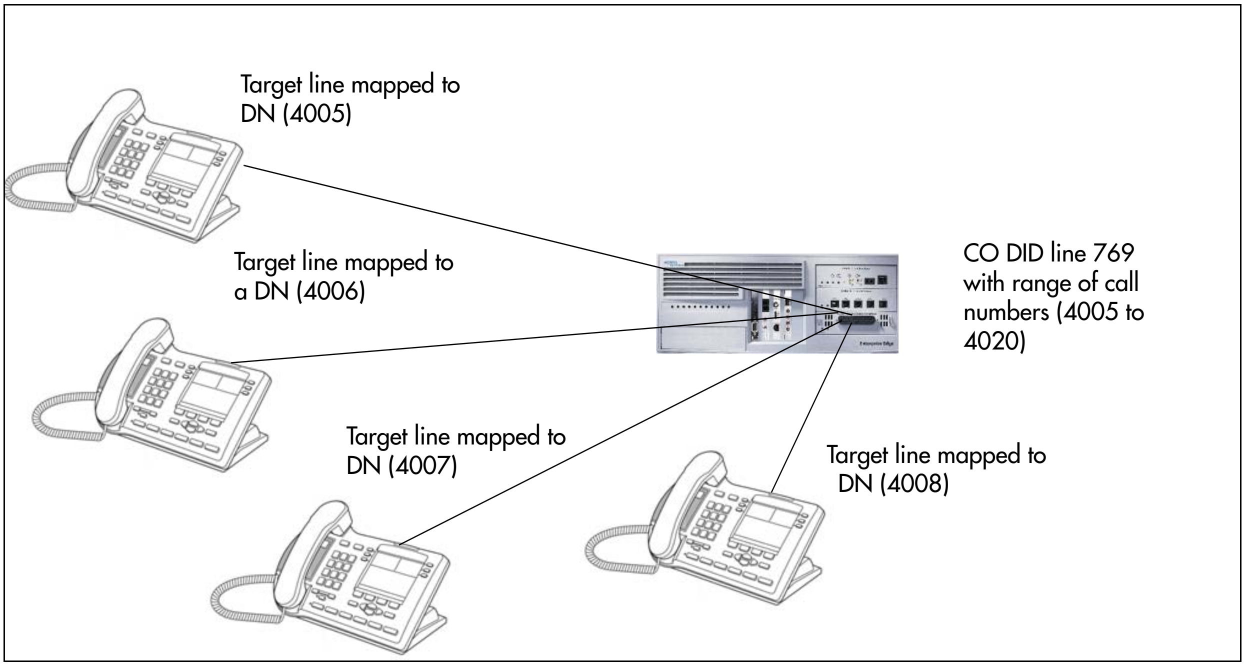
Public network: DID system

Direct Inward Dialing (DID): A call is received over the DID circuit (for example, PRI) and is preceded by a packet of information (Receive Digits) containing the number that was dialed. The BCM decodes this information and routes the call to the extension that has been programmed with the designated Target Line. The benefit to the customer is a pooled access group for incoming calls so that dedicated lines are not required to be terminated on the system for each user.

This setup allows you to assign a dedicated phone number to each telephone. The CO assigns a list of available numbers for each DID line. You can change your DN range to match these numbers, or you can use target lines to match each number with a DN.

Figure 107 illustrates an example of a DID system.

Figure 107 DID system


Target line mapped to DN (4005)
Target line mapped to a DN (4006)
CO DID line 769 with range of call numbers (4005 to 4020)
Target line mapped to DN (4007)
Target line mapped to DN (4008)

Programming:

Lines

- Assign lines as auto-answer. Note: DID lines are incoming only. PRI lines can be used for both directions.
- Configure target lines for each telephone, indicating public received number (769-4006 in the example above).

Routing

- Create line pool access code to outgoing line pool.

Telephones

- Assign target line to each telephone.
- Assign outgoing line pool to telephones.
- Set call forward no answer and call forward on busy to attendant or voice mail system, if available.

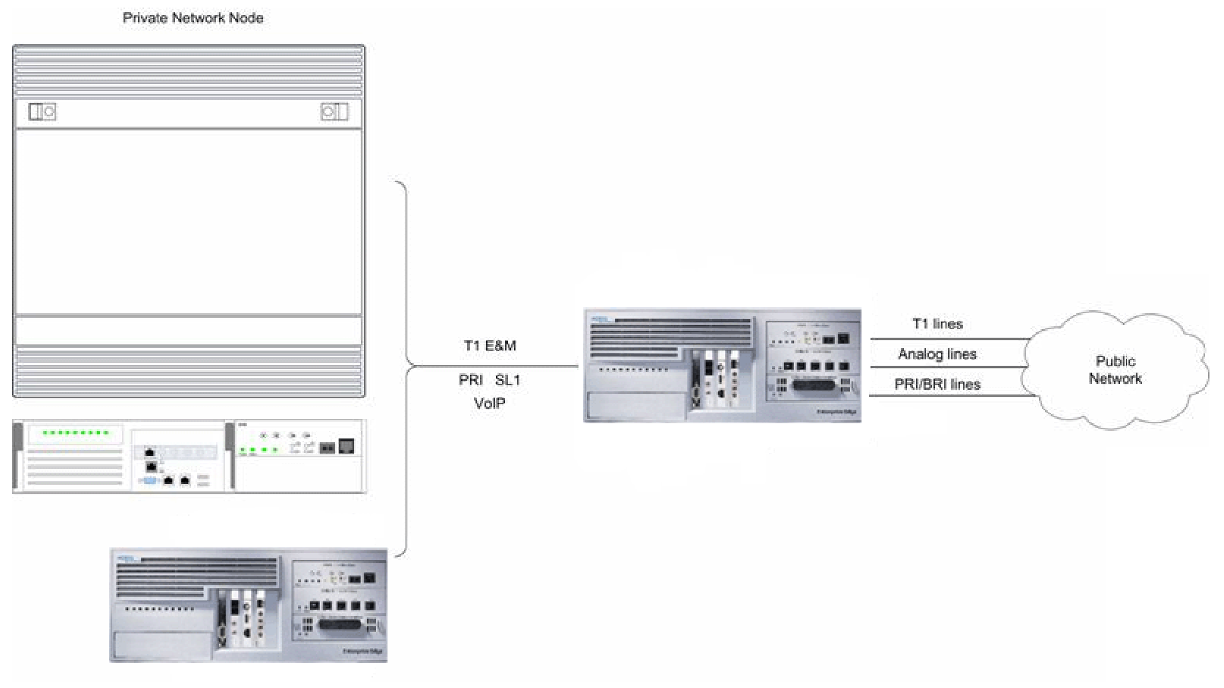
Chapter 32

Public networking: Tandem calls from private node

If your system is connected by a private network to another system that does not have PSTN line access, or which is not located within the local dialing range, you can set up a routing plan that allows the users of the private network to dial into your system, and through your system to the PSTN network. This type of call feature is referred to as tandem dialing. Refer to "Programming for tandem dialing" on page 327.

The reverse is also true. You can set up routing so that calls from the PSTN can be passed through your system and over the private network to the remote node. Also refer to "Private networking: PRI and VoIP tandem networks" on page 357.

Figure 108 Tandem dialing through a BCM to or from a private network



Programming for tandem dialing

Since incoming lines terminate within the system, you need to set up routing to pass the calls along to the required destination.

Lines:

- Set up private network lines as auto answer (if applicable).
- Put private and public lines into separate line pools.
- Assign lines to configured Remote Access Packages.

Dialing plan/Routing:

- Coordinate Dialing plan with private network node.
- Assign each line pool to a route
- Create destination codes for the private network node, and the public network, using the appropriate routes. On public route, drop the public network access code off the dial string. On the private route, drop the private network access code off the dial string.

Telephones:

- System telephones are not involved in tandem transactions. However, for calls destined for the system, ensure that the telephones have the appropriate line/line pool assignments to receive calls from both the public and private networks.

Caller access on a tandem network

In this type of configuration, there are three types of callers:

Each type of caller has a specific method of accessing the other two systems.

Callers using BCM

These callers can:

- call directly to a specific telephone
- select an outgoing line to access a private network
- select an outgoing line to access features that are available on the private network
- select an outgoing central office line to access the public network
- use all of the BCM features

Callers in the public network

These callers use the public lines to:

- call directly to one or more BCM telephones
- call into BCM and select an outgoing TIE line to access a private network
- call into BCM and select an outgoing central office line to access the public network
- call into BCM and use remote features

Callers in the private network node

These callers use private lines to:

- call directly to one or more BCM telephones
- call into BCM and select an outgoing TIE line to access other nodes in a private network
- call into BCM and select an outgoing central office line to access the public network
- call into BCM and use remote features

Chapter 33
Private networking: MCDN over PRI and VoIP

This following describes how to network BCMs together in a private network using PRI or VoIP lines with MCDN protocol.

This chapter discusses MCDN networking based on North American trunks (PRI SL-1). ETSI-QSIG private networking is configured very similarly, although network features may be supported slightly differently.

The following section describe the different aspects of MCDN private networking.

- "Using MCDN to network with a Meridian system" on page 329
- "Configuring fallback over a VoIP MCDN network" on page 342
- "Networking with ETSI QSIG" on page 344

Refer to "Private networking: Basic parameters" on page 349 for general requirements and directions for setting up non-PRI private networks.

Using MCDN to network with a Meridian system

When you connect your BCM systems through the MCDN protocol to a Meridian 1, the Meridian system manages several aspects of the network, including voice mail, auto attendant services, and system timing.

Programming note: For information about networking voice over IP (VoIP) trunks, which also can be set to use MCDN. For networks running BCM 3.6 software or later, the trunk protocol for Meridian 1 IPT connection should be set to CSE.

This section includes the follow information about setting up an MCDN network:

- "Meridian system requirements"
- "MCDN networking checklist" on page 334

For an example of an MCDN system and the BCM programming to support it, refer to "An example of a private network with Meridian 1" on page 339.

Meridian system requirements

When setting up networking with Meridian, the Meridian systems must provide the following:

- provide the correct software version to allow MCDN features. If your Meridian system administrator cannot confirm this, call your technical support center (TSC) or 1-800-4NORTEL.

  The Meridian must provide the following:

  — end-to-end signaling (option 10)

  — message center (option 46) and an IVMS link (option 35)

  - Meridian Mail link (options 77 and 85)
  - basic Attendant Console Directory features (options 40, 45, and 83)
  - ISDN PRI or ISDN Signaling link (options 145 and 146 or 145 and 147)
  - advanced ISDN features (option 148)
  - network message services (option 175)
- act as the timing master for the private network connections
- use descending mode for PRI B-channel selection
- recognize dial codes for all nodes in the network
- provide routing tables that direct incoming calls to the correct nodes on the network, including DID calls from the public network
- recognize the destination code (usually 9) that indicates a public network call, regardless of where in the network the number was dialed from

➡ **Note:** For MCDN over VoIP trunks, the Meridian uses the IPT trunk card. Both systems must have remote gateways pointed to correct system types and protocols. Refer to "Configuring VoIP trunk gateways" on page 409 for information about Remote Gateways for the BCM system.

Software requirements

These additional software packages may be required to activate all the options on the Meridian.

For a new M1 (option 81C, 61C or 51C) on X11 Rls 25, the following additional packages are required to provide the software options listed above:

- SW0059B
- SW0052D
- SW0221C
- SW0051B

For a new M1 Option 11C or 11C Mini or X11 Rel. 25, order one of the following:

- Enterprise software package
- NAS/VNS software package

Meridian MCDN call features over PRI SL-1 lines

Besides the general MCDN features described in "Using the MCDN access codes to tandem calls" on page 287, an MCDN connection with a Meridian 1 voice mail system, also provides some special call features, which are listed in Table 78.

Table 78 MCDN feature enhancements

Centralized messaging	• Message Waiting Indication
Centralized Attendant	• "Camp-on" on page 332 • "Break-in" on page 333

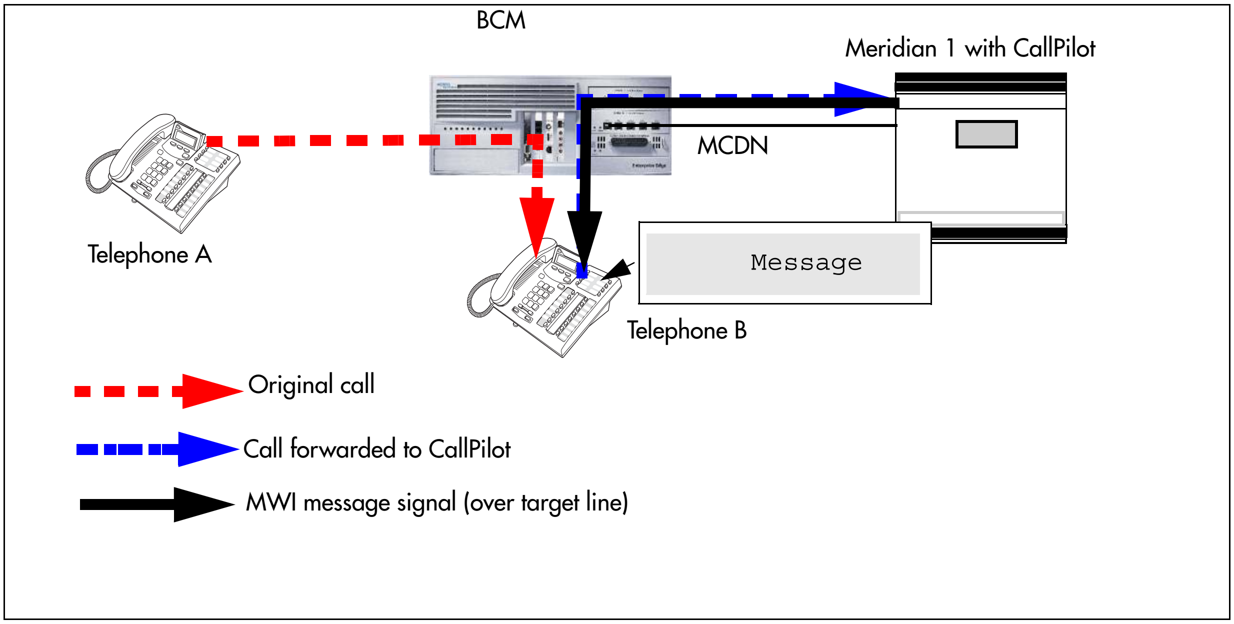
Message Waiting Indication

MWI allows the voice mail host system (Meridian 1) that is designated to receive messages to notify a target telephone on the BCM of a waiting call, using the native MCDN MWI or MIK/MCK message indicators on the Meridian telephones. This feature works for both Nortel and third-party voice mail systems. Messages are received at a centralized location, to a predetermined telephone, where they are processed and forwarded to the target telephone.

MWI allows the user to reply or call back to the message center. The procedure for retrieving messages is described in the Telephone Features Handbook.

Figure 109 demonstrates how the Meridian responds when a call is forwarded to a CallPilot mailbox.

Figure 109 Message waiting indication message


BCM
Meridian 1 with CallPilot
MCDN
Message
Telephone A
Telephone B
Original call
Call forwarded to CallPilot
MWI message signal (over target line)

Programming notes

BCM programming
To select Remote Capability for MWI on a per-loop basis for PRI: Resources, Media Bay Modules, Bus XX, Modules on bus, Module X: Remote Capability MWI = select (if M1 has MWI package, with RCAP set to MWI)
Turning on the service for IP trunks: Services, General Settings, IP Trunking: Remote Capability MWI = select (if M1 has MWI package, with RCAP set to MWI)
Telco features, VMsg Ctr Tel Numbers: • Voice Message Center 1 set to destination code plus M1 voice mail DN
Lines (target line), Telco features: • choose Voice message Center 1
System DNs, Active set DNs, Line access, Line assignment: • assign target line to each set • in target line, select VMsg

M1 programming
1. Disable the PBX D-channel associated with IPT (LD96). 2. Add MWI to the RCAP of the D-channel (LD 17 RCAP MWI) 3. Ensure the RLS ID is a minimum of 25 (RLS ID 25). 4. Re-enabled the PBX D-channel. **Note**: Package 219 is required on the Meridian PBX to allow RCAP MWI. **Note**: If IP routing is being used, you must complete this procedure on all the D-channels in the private network.

Camp-on

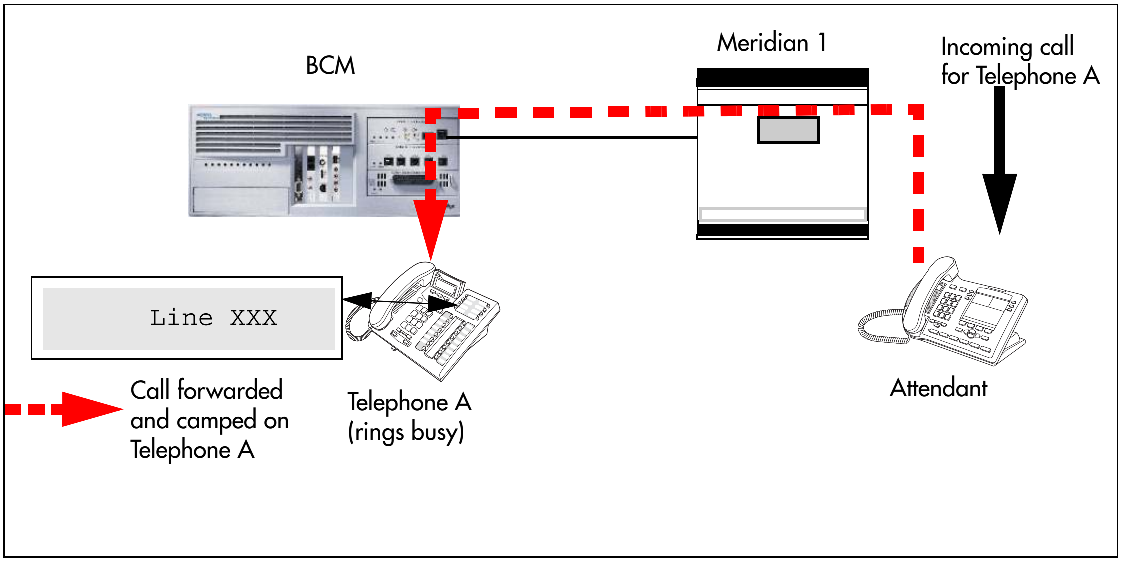
A call received by the Meridian attendant can be assigned to a telephone anywhere in the MCDN network, when the following situations are valid:

- the target telephone rings busy when the attendant calls
- no free keys on target telephone
- DND regular feature is inactive
- DND on busy feature is inactive

The target user sees that there is a call camped on the telephone. The called user can then clear a busy lines and take the call, or the user can choose to reject the call, using F814, or the user can indicate Do Not Disturb, using F85.

Figure 110 demonstrates the call path for a Meridian attendant to camp a call on a telephone in the system.

Figure 110 Camping a call

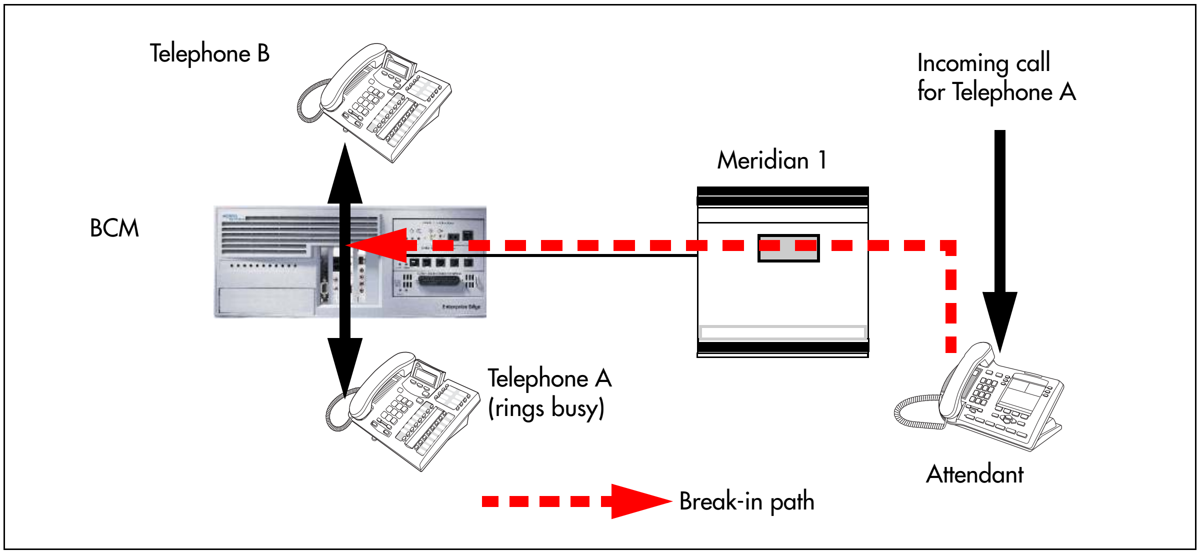

BCM
Meridian 1
Incoming call for Telephone A
Line XXX
Call forwarded and camped on Telephone A
Telephone A (rings busy)
Attendant

Break-in

The Meridian attendant can use the break-in feature to interrupt an ongoing call from a telephone in the system.

Figure 111 demonstrates the call path for a Meridian attendant to break into a call between telephones in the system.

Figure 111 Breaking into a local system call path



Break-in can occur when these situations are valid:

- Target system telephone is busy but still has a free intercom or line key.
- There is no camped call on the target telephone.

- DND on busy is turned on.
- prime set is also busy, with no free key, and with DND turned on.
- Attendant capability is high (2), and higher than either the target telephone or the caller the target telephone owner is busy with.

Only post-dial break-in is supported by MCDN:

1. Attendant dials destination number.
2. If a busy tone is heard, the attendant presses the BKI button.
   Attendant is given access to the conversation.

You can set a level of priority that will determine if a telephone will allow an attendant to break in. This is referred to as setting the Intrusion level. Use the following rules to configure the break-in feature.

- Set the Intrusion level for each telephone (under Capabilities on the DN record). Refer to “Capabilities and Preferences - Capabilities tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).

  How the intrusion hierarchy works:

  — Break-in is allowed if Attendant telephone is High and caller telephone is Medium.

  — Break-in is not allowed if Attendant telephone is Medium and caller telephone is high.

MCDN networking checklist

The following points provide a quick check for the system prerequisite settings for MCDN networking.

Select the dialing plan to be used:

- **UDP (Universal Dialing Plan)**

  — DNs on the same node are dialed directly.

  — DNs on other nodes are called by first dialing an Access Code and an ESN.

  — Each node has its own ESN.
- **CDP (Coordinated Dialing Plan)**

  — DNs on all nodes are dialed directly.

Ensure the following common programming is configured:

- **BCM Programming**

  — Configure the system DN length to match the DN length used in the rest of the private network.

  — Program the private Route: Type=Private, Dial=None.

  — Program the public Route: Type=Public, Dial=None.

  — Enable the MCDN Supplementary Services; TRO=selected, ICCL=selected, TAT=selected.

— Program telephones with a target line that specifies the system DN of the telephone in the **Private received number** field.

Note: If you have public DNs set up for your telephones that are different from the system-assigned DN, each telephone needs to use the public and private received digits on the target line.

- **Meridian 1 Programming**
  — Program the system PNI and the PNIs for the routes.
  — Program the Meridian voice mail mailboxes (if required).
  — Enable the MCDN Supplementary Services; RCAP=[ND2,TRO,MWI], NASA=YES.

Set up the specific programming the system requires for the dialing plan. Refer to the following tables.

UDP-specific programming

BCM UDP programming		
• Private Dialing Plan:	Type=UDP, HomeLoc=<three-digit prefix>	
• Private Access Code	<unique code>	
• Private DN length	<total of Private Access Code + Location Code + DN length> Example: if dialing string is 6 393 2222, then set private DN to 8	
• Program the DestCodes for the other nodes	AccessCode plus the ESN, absorb the AccessCode. Example: For AccessCode=6; DestCode=6393[Absorb=1]	

M1 UDP programming		
• Private Access Code	Overlay 86, LD 86 REQ: PRT CUST: 0 FEAT: ESN	To change Private Access Code: Overlay 86, LD 86 REQ: CHG CUST: 0 FEAT: ESN, keep pressing until you reach the AC1 prompt At the AC1 prompt, make your choice
• Check UDP programming	Overlay 90, LD 90 REQ: PRT CUST: 0 FEAT: NET TYPE: LOC LOC: press enter, all the programmed location codes are listed HLOC is the home location of the M1	
• Program UDP values to route	Overlay 90, LD 90 REQ: CHG CUST: 0 FEAT: NET TYPE: AC1 LOC: (enter a number) RLI: (enter the RLI corresponding to the route)	

CDP-specific programming

BCM CDP programming	
• Private Dialing Plan: Private Access Code <unique code>.	Type=CDP
• Private DN length	<system DN length>
• PNI	<number assigned from M1 (1-127)>
• Program the DestCodes for the other nodes	use Steering code as part of dial string

M1 CDP programming		
• PNI	LD 16, RDB - PNI in M1 programming LD 15 - Net - PNI in M1 programming set to PNI of switch	
• Distant Steering Codes	Overlay 87, LD 87 REQ: PRT CUST: 0 FEAT: CDP TYPE: DSC (Distant Steering Code) DSC: press enter (lists all DSC programmed)	
• Check RLI (Route Line Index)	Overlay 86, LD 86 REQ: PRT CUST: 0 FEAT: RLB PLI: press enter (displays all the RLIs)	
• Program new CDP value to route	Overlay 87, LD 87 REQ: CHG CUST: 0 FEAT: CDP TYPE: DSP DSC: enter number (enter common BCM system number, for example if DNs are 4XX, enter 4) RLI: enter the RLI that corresponds to the route	

VM programming with Meridian 1

If you are using the centralized voice message system from a Meridian 1 system, you require the following programming on the M1:

M1 programming in LD 17

- NASA selected
- NCRD selected

Verifying NASA is Active • Overlay 22, LD 22 • REQ: PRT • TYPE: ADAN DCH (slot number) • NASA should be selected			
If NASA is not on:	Disable the D channel • Overlay 96, LD 96 • REQ: CHG • TYPE:DISDCH	Disable the loop • Overlay 60, LD 60 • REQ: CHG • TYPE: DISL (slot number)	Program the D channel • Overlay 17, LD 17 • REQ: CHG • TYPE: ADAN • ADAN: CHG DCH (slot number) • Keep pressing enter until you get to NASA • TYPE: yes • TYPE: end
Verifying NCRD • Overlay 20, LD 20 • REQ: PRT • TYPE: TIE • CUST: 0 • Route: Enter the route defined in LD 20 • Keep pressing enter until all values are displayed. Check if NCRD is yes.		If NCRD is set to no • Overlay 16, LD 16 • REQ: CHG • TYPE: RDB • CUST: 0 • ROUT: (route number) from LD 20 • Keep pressing enter until you get NCRD and type Yes • Keep pressing enter until you get the REQ prompt again • TYPE: end	

Meridian TRO programming

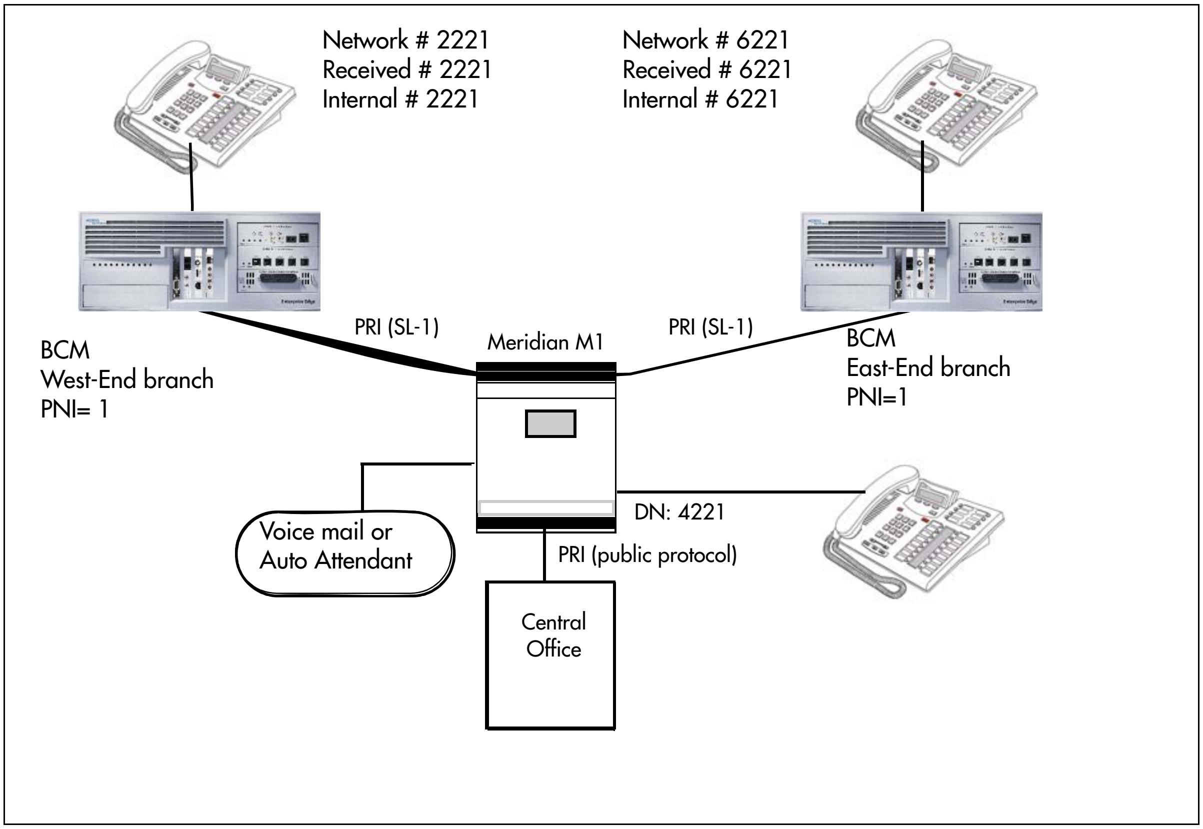
If you are using a Meridian 1 system as part of the network, you need the following programming for each system:

M1 TRO set to yes for BCM route: LD 16 TYPE: RDB Cust: xx Rout: 0-511 TRO: Yes

An example of a private network with Meridian 1

Figure 112 shows a private network composed of one central Meridian 1, and two sites with BCM systems all connected by SL-1, with MCDN activated on all sites. This example uses a coordinated dialing plan (CDP). The DNs consist of four digits. The first digit is a destination code which is specific to each system. The last three digits are unique to each telephone within that system. Refer to "Dialing plan: Private network settings" on page 317 for a description of the dialing plans available to private networks.

Figure 112 MCDN networking, with a common public network connection



This example could represent a large head office (the Meridian 1) connected to several smaller branch offices (the two BCMs). In this network, only the head office has trunks connected to the public network.

The branch offices access the public network through the PRI to the head office. This configuration allows for cost savings by consolidating the public access trunks. Users at all three locations access the public network by dialing 9, followed by the public number. For example, a user in the West End branch might dial 9-555-1212 (for a local call) or 9-1-613-555-1212 (for a long distance call). These public calls are routed to the Meridian 1 by the BCM routing table. Routing tables at the Meridian 1 will then select an appropriate public facility for the call.

Note that the Private Network Identifier (PNI) is programmed at each end of the links. The PNI identifies the BCM to the Meridian 1 system.

Routing is set up such that network calls are made by dialing a four-digit private network DN. For example, if a user in the West End branch wishes to call a user in the East End branch within the private network, they dial 6221. Figure 112 illustrates this example.

The implications on the configuration on each node to access the PSTN through one network node:

- Each node must have the Private Network Access Code set to the value 9.
- Each node must have destination codes that match the Private Network Access Code plus digits corresponding to calls terminating in the local PSTN. For example, if the Private Network Access Code is 9, the node in Ottawa would require a destination code of 91613. Similarly, Toronto would require the following destination code: 91416.

BCM module settings: Table 79 lists the module settings that are required to set up the network described in Figure 112.

Table 79 Module settings for MCDN network

West End office:		
Module programming	DTM	PRI
	Protocol	SL-1
	BchanSeq	Ascend
	ClockSrc	Primary
East End office:		
Module programming	DTM	PRI
	Protocol	SL-1
	BchanSeq	Ascend
	ClockSrc	Primary

BCM dialing plan settings: Table 80 lists the dialing plan settings that are required to set up the network described in the figure in the previous section.

Table 80 MCDN dialing plan settings

West End office:		
Dialing Plan programming	Type	CDP
	Private Network ID	1
	Private DN Length	4
	Public DN Length	7
East End office:		
Dialing Plan programming	Type	CDP
	Private Network ID	1
	Private DN Length	4
	Public DN Length	7

BCM routing information: Table 81 lists the lines and routing information required to set up the network shown in Figure 112.

Table 81 Network routing information (Sheet 1 of 2)

West End office:			
Trunk/Line Data	Line 125	Target line	
	Private Received #	2221	
Line Access	DN 2221	L125:Ring only	
	Line pool access	Line pool BlocA	
Routing Services	Private Network		Public Network
	Head Office and East end		
Route	001		002
External #	No number		No number
Use	Pool BlocA		Pool BlocA
DN type	Private		Public
Destination codes for routes to:	Head office to M1	Head office to East End	
Destination Code	4 (includes location code)	6	9
Normal route	001	001	002
Absorb	0	0	0

Table 81 Network routing information (Sheet 2 of 2)

East End office:			
Trunk/Line Data	Line 125	Target line	
	Private Received #	6221	
Line Access	DN 6221	L125:Ring only	
	Line pool access	Line pool BlocA	
Routing Services	Private Network		Public Network
	Head Office to West End		
Route	001		002
Dial out #	No number		No number
Use	Pool BlocA		Pool BlocA
DN type	Private		Public
	Head Office to M1	Head Office to West End	Call terminates at M1
Destination Code	4 (contains location code)	2	9
Normal route	001	001	002
Absorb	0	0	0

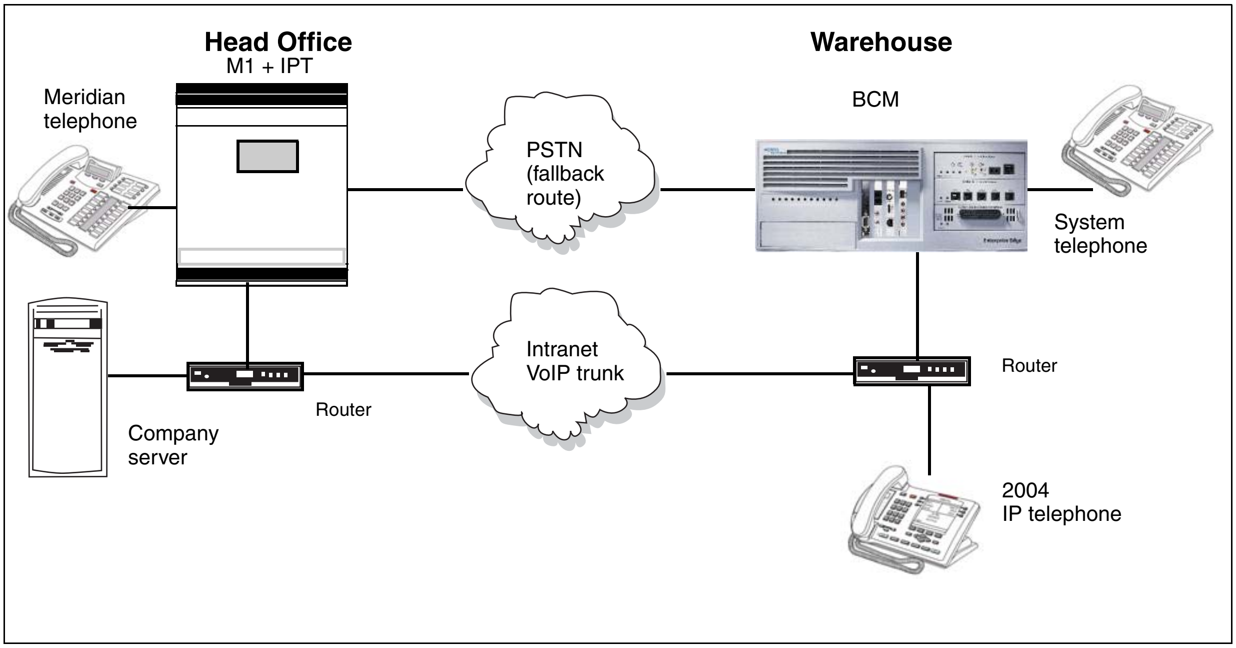
Configuring fallback over a VoIP MCDN network

The Voice over IP (VoIP) MCDN networking protocol between a Meridian 1 and one or more BCMs works the same way as it does over PRI lines. You still require the MCDN and IP telephony software keys and compatible dialing plans on all networked systems.

The one difference between MCDN over PRI and MCDN over VoIP is that the VoIP trunks require specific Remote Gateway settings, unless there is a Gatekeeper configured to route traffic on the IP network. You must also ensure that your PSTN fallback line is a PRI SL-1 line, to maintain MCDN features on the network.

Refer to Figure 113 for an example.

Figure 113 M1 to BCM network diagram



To set up the M1 in a BCM network

1 Make sure the M1 IPT meets the following requirement:

- IPT version 3.0 or newer

2 Ensure that the M1 ESN programming (CDP/UDP) is compatible. For information about this, refer to your M1 documentation.

3 On the BCM Element Manager:

- Set up outgoing call configuration for the VoIP gateway.
- Set up a remote gateway for the Meridian 1.
- Ensure the dialing rules (CDP or UDP) are compatible with the M1.
- Configure the PSTN fallback, and enable QoS on both systems.
- If target lines have not already been set up, configure the telephones to receive incoming calls through target lines.

MCDN functionality on fallback PRI lines

To enable MCDN functionality over PRI fallback lines

- Check MCDN PRI settings on the M1. For information on this, refer to the M1 documentation.
- Ensure SL-1 (MCDN) keycodes are entered on the BCM and the PRI line is set up for SL-1 protocol.

For a detailed description of setting up fallback, refer to "Setting up VoIP trunks for fallback" on page 423.

Networking with ETSI QSIG

(International systems only)

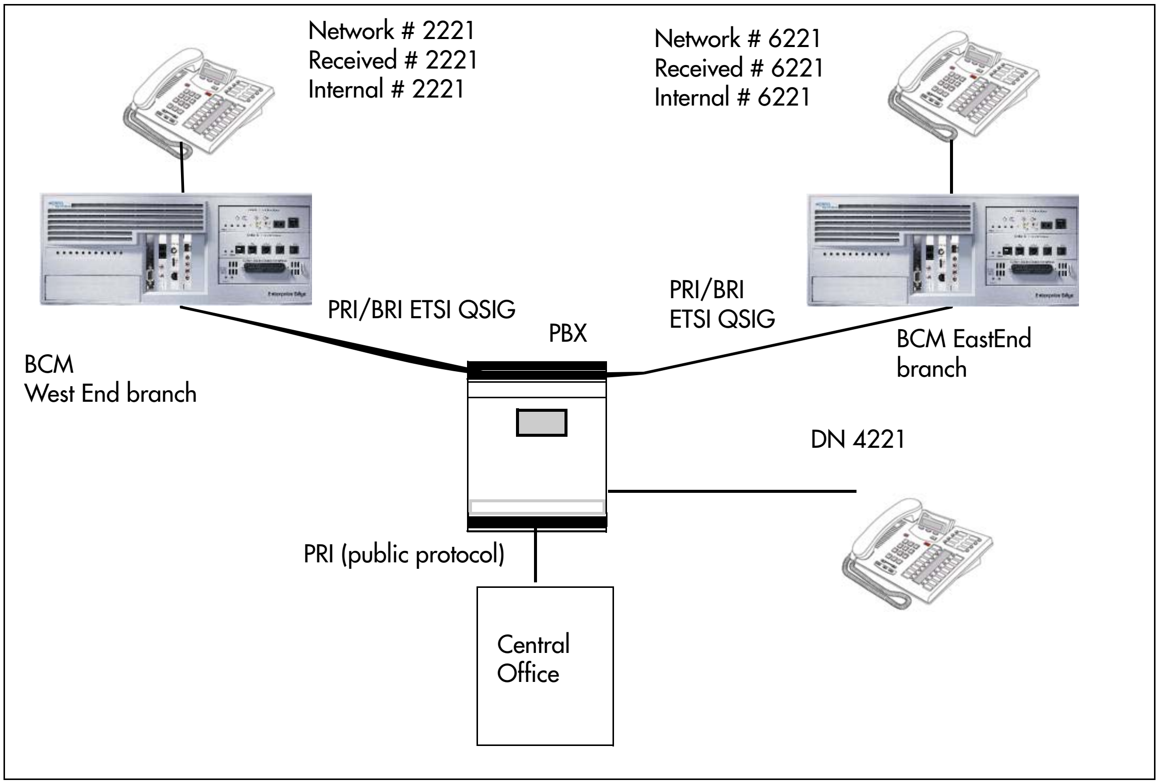
ETSI QSIG is the European standard signaling protocol for multi-vendor peer-to-peer communications between PBX systems and/or central offices.

Also refer to "Configuring ETSI Euro network services" on page 355.

Figure 114 illustrates an ETSI QSIG network. Note that this is exactly the same setup as that shown in the MCDN section for North America, in "An example of a private network with Meridian 1" on page 339, which describes PRI SL-1 networking. The exception in the configuration is for the hardware configuration because the trunk lines are different. The hardware programming for ETSI QSIG is described below the following diagram. All other configurations are the same as those shown in the MCDN section for North America, in "Using MCDN to network with a Meridian system" on page 329.

Note: Features for ETSI Q.sig are basic compared to MCDN. Only basic call and calling number is supported as opposed to the many MCDN features.

Figure 114 ETSI QSIG networking



Settings for some of the hardware parameters for the ETSI QSIG networking example shown above are as follows.

West End office:		
Hardware programming	DTM/BRIM	PRI/BRI
	Protocol	ETSI QSIG
	BchanSeq	Ascend (PRI only)
	ClockSrc	Primary

East End office:		
Hardware programming	DTM/BRIM	PRI/BRI
	Protocol	ETSI QSIG
	BchanSeq	Ascend (PRI only)
	ClockSrc	Primary

Chapter 34

Private networking: Using shared line pools

Using shared line pools is a powerful and efficient way to create a coordinated dialing plan for a small network. If the BCM systems are close to each other geographically, you can conserve resources by not duplicating long-distance access. For example, system A, B, and C are all within the same area code. System A has a line pool to Santa Clara, System B has a line pool to Montreal, and system C has a line pool to Miami. A BCM user in system A can reach Miami by calling system C and using their line pool to Miami.

To simplify access between BCM systems, all line pools that go to the same destination should have the same line pool access code. For example, system A and system B both have a line pool to Ottawa. You can configure both systems with the same line pool access code for the Ottawa line pool.

A dialing plan similar to the one in Figure 115 allows you to create a company directory that uses line pool access codes and unique DNs of a uniform length.

In this example, the person on system A at telephone 234 can press an intercom button and dial 7434.

This means that telephone 234 has dialed the line pool access code of the trunk to system C, and will receive the dial tone of system C. The digits 434 then map to the Private received number 434, and ring telephone 434 with an appearance of the associated target line.

Figure 115 Network example using shared line pools

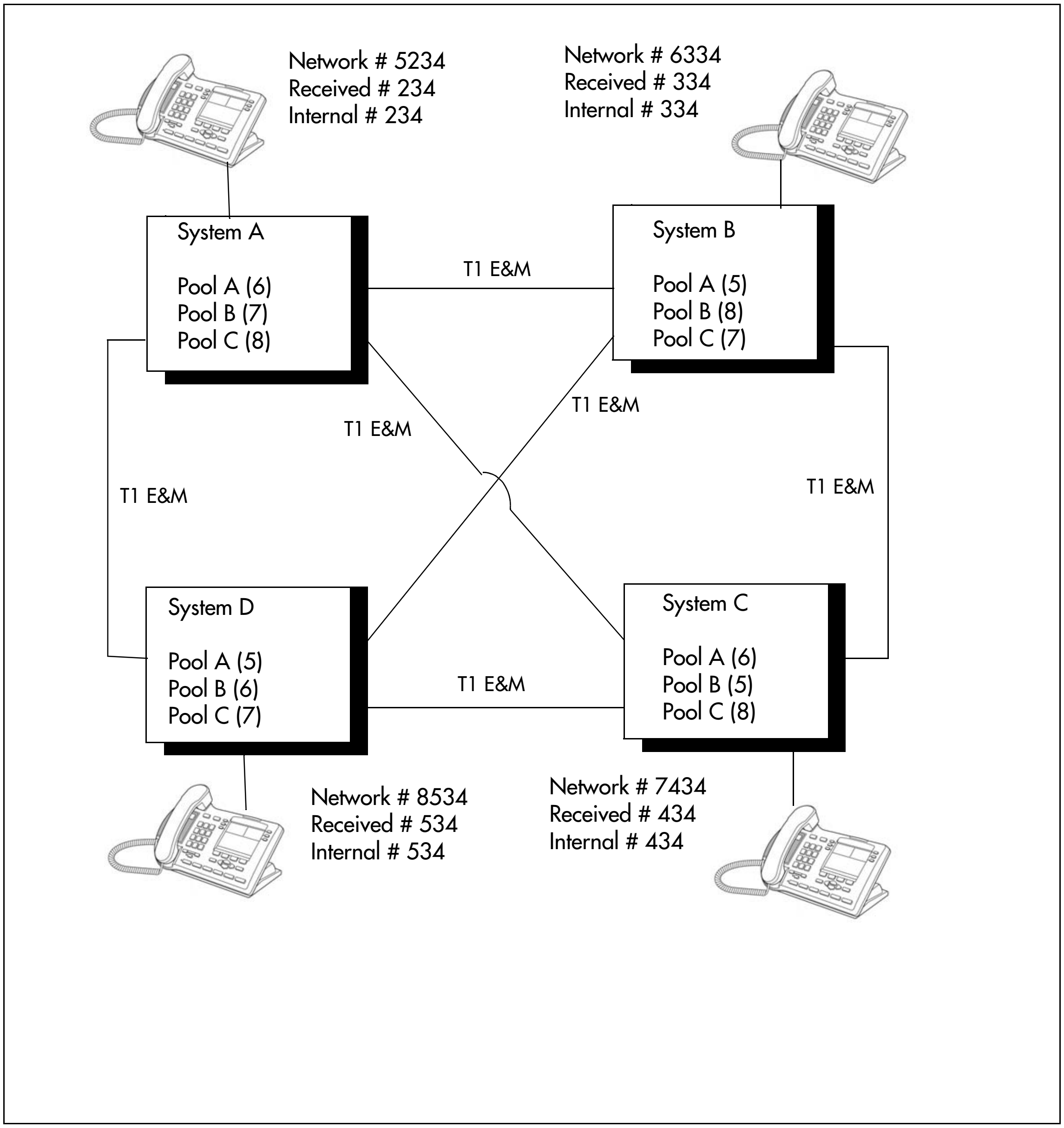


Table 82 shows the system coding for each system to set up a line pool-based coordinated dialing plan.

Table 82 Creating a coordinated dialing plan using line pools

Route from System A to: System		B	C	D
Dialout:	5234	6334	7434	

Chapter 35

Private networking: Basic parameters

The following provides an overview of the values in the system that affect private networking, including:

- “Private networking protocols” on page 349
- “Keycode requirements” on page 349
- “Remote access to the network” on page 350
- “Other programming that affects private networking” on page 350
- “Types of private networks” on page 350

Private networking protocols

These are the protocols that the BCM supports for private networking:

- PRI: ETSI QSIG, MCDN, DPNSS
- BRI: ETSI QSIG
- T1: E&M
- VoIP: MCDN

BCM systems can be networked together using TIE lines, or E&M, connections. Larger networks, or networks that are geographically spread out, can be chained together through faster PRI SL-1 connections or with voice over IP (VoIP) trunk lines. SL-1 lines and VoIP trunks also offer the opportunity to use the MCDN protocol, which provides enhanced trunking features and end-to-end user identification. If a Meridian 1 is part of the MCDN network, the network can also provide centralized voice mail and auto attendant off the Meridian.

MCDN note: MCDN networking requires all nodes on the network to use a common Universal Dialing plan (UDP) or a Coordinated Dialing Plan (CDP). Refer to “Dialing plan: Public network,” on page 311 and “Dialing plan: Private network settings,” on page 317.

Keycode requirements

Keycodes are required to activate the protocols that are used to create private networking, including:

- IP trunks, if you want additional IP trunks
- an MCDN keycode, if you want to use the MCDN protocol between the systems

You must purchase and install these keycodes before you can create any of the networks described in this chapter. Consult with your Nortel distributor to ensure you order the correct keycodes for the type of network you want to create.

Remote access to the network

Authorized users can access TIE lines, central office lines, and BCM features from outside the system. Remote users accessing a private network configured over a large geographical area, can potentially also place long-distance calls through the network and avoid toll charges. Also refer to "Call security and remote access" on page 441.

➡ **Note:** You cannot program a Private DISA DN or Private Auto DN to a VoIP trunk, as they act as auto-answer trunks from one private network to the next. However, you can configure VoIP line pools with remote access packages so that callers can access telephones or the local PSTN on remote nodes on a tandemed network that use VoIP trunks between systems.

Other programming that affects private networking

Besides the line programming, these links connect to other programming that affects or is affected by private networks.

- "Dialing plan: System settings," on page 303 (Received Number Length)
- "Defining trunk module types and settings" on page 106
- "Configuring lines" on page 147
- "Configuring lines: Target lines," on page 177
- "Dialing plan: System settings," on page 303
- "Dialing plan: Routing and destination codes," on page 289
- "Call security: Restriction filters," on page 457
- "Call security: Remote access packages," on page 463
- "Configuring CLID on your system," on page 235
- "Line Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600) (Private OLI)

Types of private networks

There are several ways you can create private networks. Configuration can be based on such things as cost of trunks, proximity of network nodes, size of the private network, and business requirements for communications.

VoIP-based networking also requires an understanding of IP features such as codecs, jitter buffers, Quality of Service (QoS) function, and silence suppression.

The services provided within networks is based on the type of trunks and the protocols assigned to the trunks. All trunks within the network should be running the same protocols, to provide a consistent look and feel to the users.

These are the main types of private networking, listed from the simplest to the more complex PRI/ETSI and VoIP routing using MCDN protocols:

- "Private networking: Using destination codes," on page 373

- “Private networking: Using shared line pools” on page 347
- “Private networking: PRI Call-by-Call services,” on page 377
- “Private networking: PRI and VoIP tandem networks,” on page 357
- “Private networking: MCDN over PRI and VoIP,” on page 329
- “Private networking: DPNSS network services (UK only),” on page 365

Chapter 36

Private networking: MCDN and ETSI network features

If the MCDN protocol is added to a PRI SL-1 or VoIP private network, the network provides additional network management features, as well as allowing centralized voice mail features to be available to all nodes on the network.

ETSI lines (UK profile) also have network features available from the central office that can be enabled or disabled.

The following describes the different aspects of SL-1 and MCDN private networking.

- "Configuring MCDN network features" on page 353
- "Configuring ETSI Euro network services" on page 355

Configuring MCDN network features

When you connect your BCM systems through PRI SL-1 or VoIP trunks, and activate the MCDN protocol, your network provides a number of network call features. You can use this protocol to network other BCM systems, Norstar systems, Meridian 1 systems, and Succession systems.

Table 83 lists the MCDN features that are provided by all SL-1/VoIP networks where MCDN is active. The features affect call redirection and trunking functions.

Table 83 MCDN network features

Centralized messaging	"Configuring Network Call Redirection Information" on page 353 (NCRI)
Centralize trunking	"ISDN Call Connection Limitation" on page 354 (ICCL) "Trunk Route Optimization (TRO)" on page 354 (TRO) "Trunk Anti-tromboning (TAT)" on page 354 (TAT)

Configuring Network Call Redirection Information

NCRI provides call information in the network when calls are redirected from one system to another. NCRI builds on the following BCM features:

- External Call Forward
- Call Transfer
- Call Forward

ISDN Call Connection Limitation

The ICCL feature piggybacks on the call initiation request and acts as a check at transit PBX points to prevent misconfigured routes or calls with errors from blocking channels.

To configure ICCL

1. To access the Private Network panel, select **Configuration > Telephony > Dialing Plan**.
2. Locate the Private Network/MCDN subpanel.
3. Select the Network ICCL check box.
4. To access the Telephony Resources panel, select **Configuration > Resources**.
5. From the Modules table, select the required module.
6. Locate the Details for Module subpanel.
7. Click the Trunk Module Parameters tab.
8. Enter the Maximum transits in the Maximum transits field.

Trunk Route Optimization (TRO)

TRO finds the most direct route through the network to send a call between nodes. This function occurs during the initial alerting phase of a call.

To enable TRO

1. Click **Configuration > Telephony > Dialing Plan**.
2. Locate the MCDN subpanel.
3. Select the TRO check box.

Trunk Anti-tromboning (TAT)

TAT is a call-reroute feature that works to find better routes during a transfer of an active call. This feature acts to prevent unnecessary tandeming and tromboning of trunks.

Note: TAT is not applicable for alerting calls.

To enable TAT

1. Click **Configuration > Telephony > Dialing Plan.**
2. Locate the MCDN subpanel.

3 Select the **TAT** check box.

Configuring ETSI Euro network services

If your system has ETSI ISDN BRI/PRI lines, you can activate the malicious call identification (MCID) and Network Diversion features. Advice of Charge-End of Call (AOCE) is active if your service provider has activated that service on the line.

When the features are activated, users can:

- display a call charge
- redirect calls over the ETSI ISDN BRI/PRI line to the outside network
- tag malicious calls

Advice of Charge-End of Call (AOCE) — AOCE is a supplementary service available from your service provider on ETSI ISDN BRI/PRI links. This feature allows the BCM user to view the charges for an outgoing call once the call completes. This information is also reported to the Call Detail Reporting Application. The information can be provided in currency or charging units, depending on how the feature is set up by your service provider.

To invoke the feature, the user presses **FEATURE 818**.

To enable MCID and network diversion

1 To access the Private Network panel, select **Configuration > Telephony > Dialing Plan**.

2 Locate the ETSI subpanel.

Select the check boxes of the required options.

Table 84 lists the possible values for ETSI.
The **Description** column of the table describes the feature and how the user activates each feature from their telephone.

Table 84 ETSI network values

Attribute	**Values**	**Description**
Netwrk Diversion	<check box>	Allows calls to be redirected to an outside network.
MCID	<check box>	Malicious Call Identification When selected, the called party can use **FEATURE 897** to request the network to record the identity of an incoming call. including: • called party number • calling party number • local time and date of the activity • calling party sub-address, if provided by the calling user
MCID note	➡	The feature code must be entered within 25 seconds of the caller hanging up. (A 25-second busy tone occurs.) If the called party hangs up first, there is no opportunity to use the feature. **Note**: The call identification comes from your service provider, not the BCM. You must have the service activated by the CO before the feature is active for the user, regardless of the setting in this field.

Chapter 37

Private networking: PRI and VoIP tandem networks

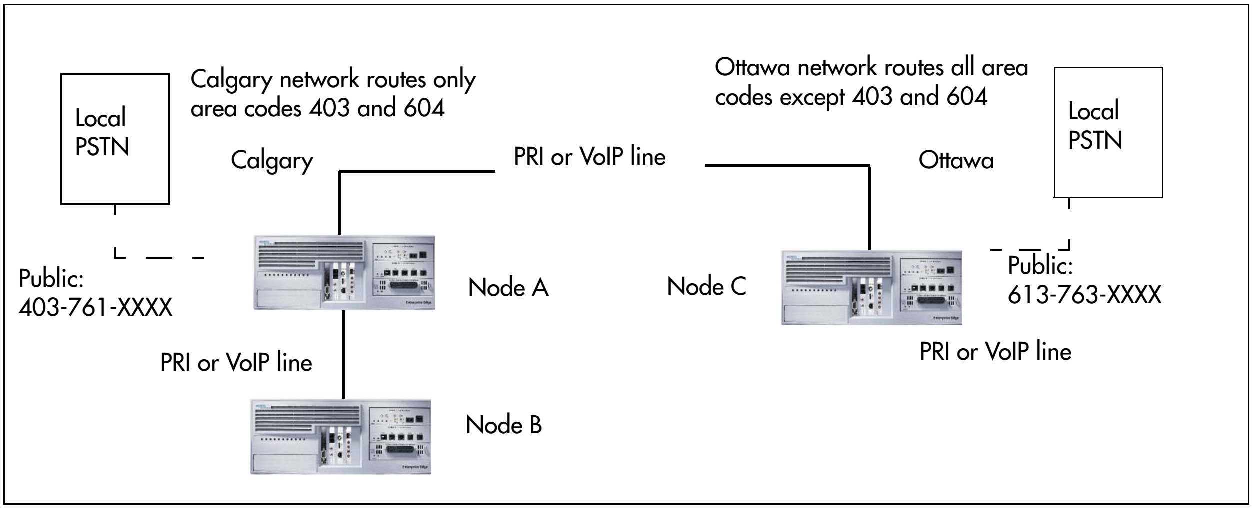
PRI trunks and VoIP trunks can be used to create a private network between other BCMs. This tandem network provides you with the benefits of end-to-end name display and toll-free calling over the PRI or VoIP private link. Each BCM becomes a node in the network.

This section includes the following information about tandem networks:

- "Routing for tandem networks" on page 357
- "Routing calls through a tandem network" on page 358
- "Using VoIP to tandem systems" on page 361

Figure 116 demonstrates a tandem configuration.

Figure 116 Private tandem network of BCMs



Also refer to "Using VoIP to tandem systems" on page 361 for other examples of tandem systems using VoIP trunks.

Routing for tandem networks

In this type of network, each Business system node is set up to route calls internally as well as to other nodes on the system. Each node must have a unique identification number, which is determined by the type of dialing plan chosen for the network.

VoIP trunks require local gateway configuration and either remote gateway or Gatekeeper configurations that identify the other nodes in the network.

If the node is also connected to the public network, the usual routing is required for that connection.

The following tables show the routing tables for Node A and Node C for external and internal terminating calls.

Table 85 Node A destination code table, external termination

Route	Absorb length	Destination code (public DNs)
4 (PSTN)	1	91604
3 (Node B)	0	91403762 (Node B)
4 (PSTN)	1	9140376* (not internal network)
4 (PSTN)	1	914037* (not internal network)
4 (PSTN)	1	91403* (not internal network)
4 (PSTN)	1	9* (not internal network)

* This wild card represents a single digit.

Table 86 Node A destination code table, internal termination

Route	Absorb length	Destination code (private DNs)
3 (Node B)	0	392 (Node B)
5 (Node C)	0	393 (Node C)

Table 87 Node C destination code table, external termination

Route	Absorb length	Destination code (Public DNs)
3 (Node B)	0	91613764 (Node D)
3 (Node B)	0	91613766 (Node F)
4 (PSTN)	1	9161376* (not internal network)
4 (PSTN)	1	916137* (not internal network)
4 (PSTN)	1	91613* (not internal network)
4 (PSTN)	1	9161* (not internal network)
4 (PSTN)	1	916* (not internal network)
4 (PSTN)	1	91* (not internal network)
4 (PSTN)	1	9 (not internal network)

Table 88 Node C destination code table, internal termination

Route	Absorb length	Destination code (Private DNs)
5 (Node A)	0	391 (Node A)
5 (Node A)	0	392 (Node B)

Routing calls through a tandem network

The following provides a step-by-step description of how calls network through a tandem network.

- "Calls originating from the public network" on page 359

- "Calls originating in the private network" on page 360

Calls originating from the public network

Table 89 describes how each node handles calls originating from the public network into the system.

Table 89 Call originating from the public network to a tandem network (Sheet 1 of 2)

Received	Destination	Description
Node A	Node A	User in Calgary dials 761-xxxx number Incoming interface: Public DN type: Public Node A receives the call and identifies it as terminating locally. Uses target line to route call (Public received #). Destination: Local (target line)
Node A	Node B	User in Calgary dials a 762-xxxx number DN type: Public Node A receives it and identifies it as being for node B. Uses private trunk to route it to B. Incoming interface: Public Destination: Remote Node Outgoing interface: Private Node B receives the call and identifies it as terminating locally. Uses target line to route call (Private received #). Incoming interface: Private Destination: Local (target line)
Node A	Node C	An external user in Calgary dials a 761-xxxx number which is answered with DISA. Incoming interface: Public DN type: Public Destination: Local (DISA DN) User enters a CoS password and a private DN for Node C 6 + 393-xxxx DN type: Private Node A receives it and identifies it as being for C. Uses the private trunk to route the call to C. Incoming interface: (DISA user) Destination: Remote node Node C receives the call and identifies it as terminating locally. Uses target line to route call. (Private received #) Incoming interface: Private Destination: Local (target line)

Table 89 Call originating from the public network to a tandem network (Sheet 2 of 2)

Received	Destination	Description
Node A	Ottawa PSTN	An external user in Calgary dials a 761-xxxx number which is answered with DISA. User enters a CoS password and an Ottawa public network number. Incoming interface: Public DN type: Public Destination: Local (DISA DN) Node A receives it and identifies it as being for C. Uses the private trunk to route the call to C. Incoming interface: Local (DISA user) Destination: Remote PSTN Node C receives the call and identifies it as a public number and routes it out over the local PSTN. Incoming interface: Private Destination: Local PSTN

Calls originating in the private network

Table 90 describes how each node handles calls originating in the public network.

Table 90 Calls originating from the private network within a tandem network (Sheet 1 of 2)

Received	Destination	Description
Node B	Node B	DN is internal, therefore no trunk routing is required. Incoming interface: Intercom DN type: Local Destination: Local
Node A	Ottawa PSTN	User in Node A dials the private network access code for Node C, followed by an Ottawa public number. Incoming interface: Intercom DN type: public Destination: Remote PSTN Node C receives the call and identifies it as being for the public network. Node C routes the call over the local public network. Incoming interface: Private DN type: Public Destination: Local PSTN
Node B	Calgary PSTN	User on Node B dials a public DN. Node B recognizes it as being the responsibility of Node A and uses private trunk to route the call to A. Incoming interface: Intercom Destination: Remote node Node A receives the call and identifies it as being for the public network. Node A routes the call over the local public network. Incoming interface: Private Destination: Remote PSTN

Table 90 Calls originating from the private network within a tandem network (Sheet 2 of 2)

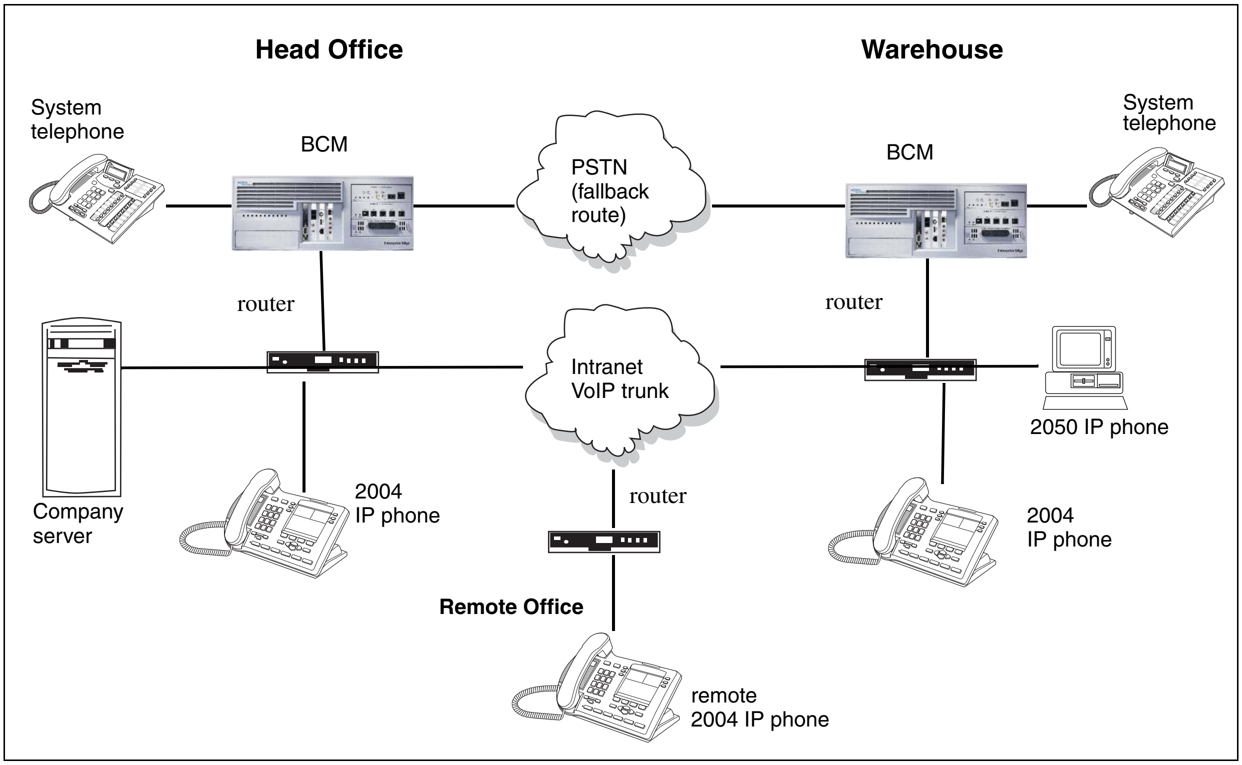
Received	Destination	Description
Node B	Node A	User in Node B dials a private DN for a user on A. DN type: Private Node B recognizes it as being for Node A. Uses the private trunk to route the call the call to A. Incoming interface: Intercom Destination: Remote node Node B receives the call and identifies it as terminating locally. Uses target line to route call. (Private received #) Incoming interface: Private Destination: Local (target line)
Node B	Node C	User on Node B dials a private DN for a user on C. DN type: Private Node B recognizes it as being the responsibility of Node A and routes the call over the private trunk to A. Incoming interface: Intercom Destination: Remote node Node A receives it and identifies it as being for C. Uses IP trunk to route call to C. Incoming interface: Private Destination: Remote node Node C receives the call and identifies it as terminating locally. Uses target line to route call. (Private received #) Incoming interface: Private Destination: Local (target line)

Using VoIP to tandem systems

You can connect multiple offices with BCMs across your company Intranet. This installation allows for CallPilot to direct calls throughout the system or for one system to support voice mail for the network. Full toll bypass occurs through the tandem setup, meaning that any user can call any DN without long distance charges being applied. Users have full access to system users, PSTN connections.

Figure 117 shows a multiple-BCM network. The network diagram shows two BCMs, but additional base units can be added.

Figure 117 Multiple BCMs network diagram



To set up a network of BCMs

1 Ensure that the existing network can support the additional VoIP traffic.

2 Coordinate a Private dialing plan between all the systems.

3 On each BCM:

- Set up outgoing call configuration for the VoIP gateway.
- Set telephones to receive incoming calls through target lines.
- Configure the PSTN fallback and enable QoS on both systems.

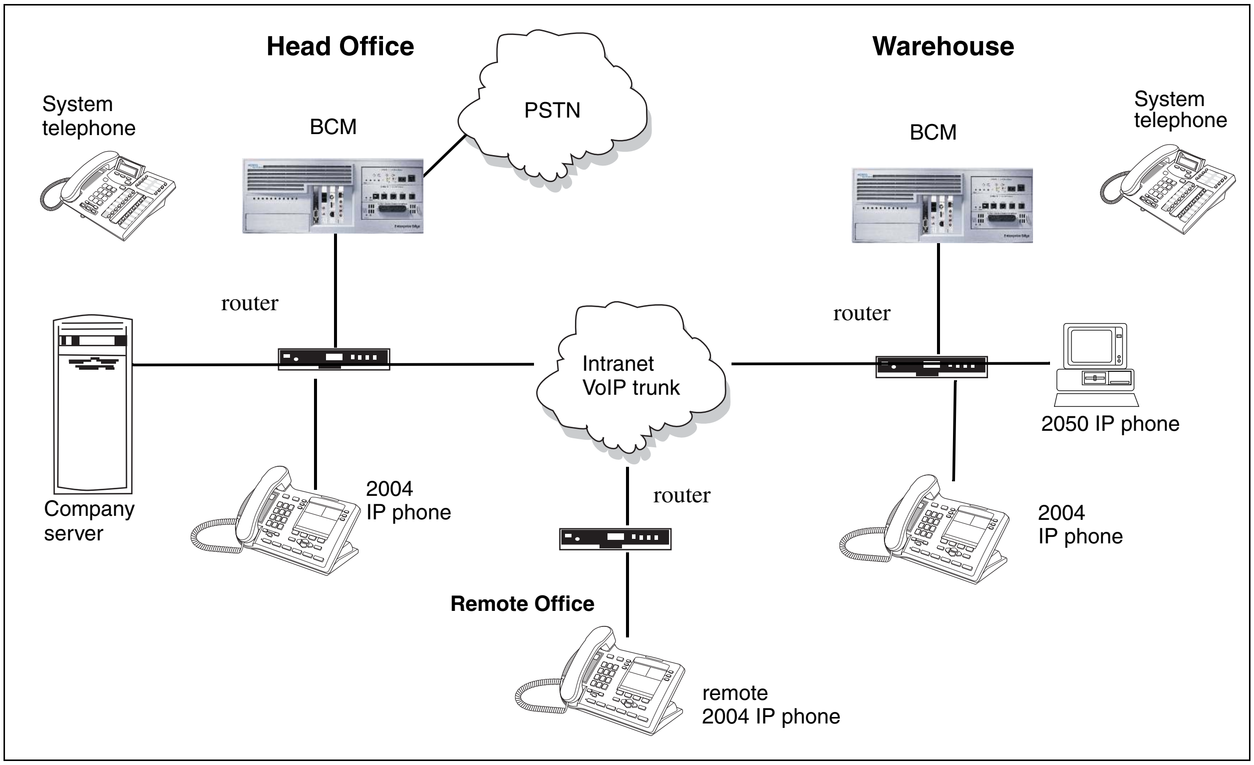
This system uses fallback to PSTN so calls can be routed across the PSTN connection if VoIP traffic between the BCMs becomes too heavy.

If only one of the BCMs in a network has a line to the PSTN network, all public calls from other systems are funneled through the system with the PSTN connection and all communication between the systems occurs over VoIP trunks. To facilitate this system, you need to ensure that the destination codes on the non-PSTN system point to the system connected to the PSTN, and then, to the PSTN. On the PSTN-connected system, the system and destination codes must be configured to recognize and pass public calls from the other system out into the PSTN network. Since the receiving PSTN sees the calls as remote dial-ins, ensure that the correct remote access packages have been established for the VoIP trunks.

This also means that if the VoIP trunks are inaccessible between the systems, there is no provision for a fallback route.

Figure 118 shows an example of routing all public calls through one BCM.

Figure 118 Routing all public calls through one BCM



Chapter 38

Private networking: DPNSS network services (UK only)

Programming note: software keys are required to enable DPNSS 1. DPNSS 1 is not available on all profiles.

The following features are available and can be programmed over DPNSS lines:

- Diversion ("Using the diversion feature" on page 365)
- Redirection ("Using the Redirection feature" on page 366)
- "Executive intrusion, Intrusion protection level" on page 367
- "Call offer" on page 367
- "Route optimization" on page 368
- "Loop avoidance" on page 368
- MWI is discussed with central voice mail setup ("Configuring centralized voice mail" on page 385)

Using the diversion feature

Diversion is a DPNSS 1 feature for BCM that allows users to forward their calls to a third party on the DPNSS 1 network. This feature is similar to call forward on BCM, but takes advantage of the broader capabilities of DPNSS.

There are five variations of Diversion: Call Diversion Immediate, Call Diversion On Busy, Call Diversion On No Reply, Bypass Call Diversion, and Follow-me Diversion. These variations are described below:

- Diversion Immediate diverts all calls to an alternate telephone. This function is programmed by the user at their telephone.
- Diversion On Busy diverts all calls to an alternate telephone when a telephone is busy. This feature is programmed in the Element Manager.
- Diversion On No Reply diverts calls that go unanswered after a specified amount of time. This feature is programmed in the Element Manager.
- Bypass Call Diversion overrides all call forward features active on a telephone over a DPNSS line. An incoming call to the telephone will not be forwarded; instead, the telephone will continue to ring as if call forward were not active. This feature is used to force a call to be answered at that location. Bypass Call Diversion is a receive-only feature on BCM, and cannot be used from a BCM telephone.
- Follow-me Diversion is also a receive-only feature. It allows the call forwarded destination to remotely change the BCM call forwarding programming (Call Forward All Calls (CFAC) feature) to a different telephone.

➡ **Note:** BCM CFAC must be active and the destination set/PBX system must support the feature.

For example, user A forwards all calls to telephone B, a temporary office. Later, user A moves on to location C. The user does not have to be at telephone A to forward calls to location C. Using telephone B and Follow-me Diversion, the user can forward calls from A to location C.

Follow-me diversion can be cancelled from the forwarded location.

- Diversion on Busy and Diversion on No Reply cannot be cancelled from the forwarded telephone. These are programmable only by an installer and not by the user.
- If multiple telephones are programmed to take a call, the first telephone to respond will act. All other telephones responding are ignored. Therefore, if the first telephone to respond has Diversion enabled, this feature will be invoked.

Restrictions by telephone type

- all variations supported on BCM digital and IP telephones
- ATA2/ASM8+—all variations supported on an ATA
- ISDN—all variations supported on ISDN telephones, except Diversion on Busy and CFWD Busy

Setting Diversion

You set Diversion for DPNSS in the same way as call forward. You will need to enter the end DN when prompted. You may also need to include the DPNSS 1 routing number.

DPNSS to Embark connections

DPNSS lines connected to an Embark switch perform call redirection/diversion using the Call Forward feature to create a tandem link back to the switch. Since this is different from other switches, you must select the type of switch DPNSS will be connecting to when you do module programming. Refer to "Configuring the trunk module to line type" on page 107.

Before you program Call Forwarding ensure that:

- Both real channels and virtual channels are provisioned.
- Destination or line pool codes are programmed for the DPNSS to Embark link.

Also, during programming for Call Forward No Answer and Call Forward on Busy, when you enter the **Forward to:** digits, the system does a validation check with the switch on the number. (select **Configuration > Telephony > Sets, All DNs panel, Line Access tab**, double-click the required field to enter the DN).

Using the Redirection feature

Redirection is a DPNSS 1 feature similar to BCM Transfer Callback. Redirection lets a call awaiting connection, or re-connection, be redirected by the originating party to an alternate destination after a time-out period. Failed calls can also be redirected. Priority calls are not redirected.

Diversion on No Reply feature takes precedence over Redirection.

Restrictions by telephone type

- For telephones with single line displays, the # key acts as `MORE` and the * key acts as `VIEW`
- ATA2/ASM8+—not supported
- ISDN—all variations supported on ISDN telephones

Setting redirection

The timer used for the network Callback feature is also used for redirection.

Executive intrusion, Intrusion protection level

Executive Intrusion (EI) is a DPNSS 1 feature that allows an operator, or other calling party, to intrude on a line when it is busy. An example of the use of this feature is to make an important announcement when the recipient is on another call.

EI is implemented on the BCM using Intrusion protection level (IPL). IPL has four settings, from None to High. A telephone set has the ability to break-in when the other telephone set has a lower IPL. The default setting is None and a setting of High prevents intrusion.

Restrictions by telephone type

- ATA2/ASM8+—supported
- ISDN—not supported

To program IPL

1. To access the All DNs panel, select **Configuration > Telephony > Sets**.
2. On the panel, locate and click the **Capabilities and Preferences** tab.
3. Select the DN of the telephone set being programmed.
   The Details subpanel for that DN appears in the lower portion of the panel.
4. Click the **Capabilities** tab.
5. Locate the Intrusion protection level and select the required option from the drop-down menu.

Call offer

Call Offer over DPNSS 1 allows a calling party to indicate to the wanted party that there is an incoming call available, even though there is no answer button available to present the call on the telephone.

Restrictions by telephone type

- model 7000 telephone — associated LED or LCD flashes, and a tone is heard
- ATA2/ASM8+—Call Offer is supported as a Camp On feature, and a tone is heard
- ISDN—not supported

Note the following general conditions and restrictions:

- DND on busy must be programmed as N (**DN ##/Capabilities**) for a telephone to accept Call Offer.
- If CF on busy is programmed for the telephone, Call Offer is not accepted.
- The target line for the telephone must be set to: If **busy: busy tone**, which is the default. Refer to "Configuring lines: Target lines" on page 177.
- Call Offer does not work if sent over Manual answer lines. It is recommended that the lines be left at the default: **Auto**.

Note: Forward on Busy takes priority over DND on Busy. Call Offer cannot be accepted by putting an active call on hold.

Route optimization

Route Optimization is a DPNSS 1 feature for BCM that allows calls to follow the optimum route between two end PBXs. This allows efficient use of network resources.

There is no system programming required for the feature when BCM is working as a terminating PBX system. However, BCM must have a private access code programmed that maps to a valid destination code or line pool code on DPNSS lines. Further, Allow redirect must be set to selected. For more information, see "Capabilities and Preferences - Capabilities tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Loop avoidance

To set Loop avoidance during hardware configuration

1. Select **Configuration > Resources > Telephony Resources**.
2. Select the **DPNSS** module.
3. Type a value (0-25) in the Maximum transits box.
   The default value is 25.

Private networking with DPNSS

(International only)

DPNSS supports the Universal Dialing Plan (UDP), an international standard for sending and receiving private numbers over networks. The UDP requires that a dialing number includes the following:

- a Private Access Code, programmed into the system as part of the destination code table to prevent conflicts with the internal numbering system. (**Access Codes**)
- a Home Location Code (HLC) assigned to each PBX system, and configured as part of the destination code (a maximum of seven digits). For each HLC, a destination code must be programmed in the system. (Dialing plan, UDP, Location code)
- a Directory Number (DNs) assigned to each extension as a line appearance. The DN appears as the last string segment in a dialed number. In the number 244-1111, 1111 is the DN.

A typical Private Number, using a private access code and dialed from another site on the network, appears below.

Private Access Code	+ Home Location Code	+ Directory Number	= Calling Party Number
6	+ 848	+ 2222	= 6-848-2222

In this networking example, a private network is formed when several systems are connected through a Meridian M1 and a terminating BCM system. Each site has its own HLC and a range of DNs. Figure 119 illustrates this example.

Calls are dialed and identified to the system as follows:

- To reach a telephone inside the Private Network, at the BCM site, the user dials the DN of choice.
- To reach a telephone inside the Private Network, from another site, the user dials HLC + DN.
- To reach a telephone outside the Private Network, the user dials an Access Code + HLC + DN

  Each node has its own destination (dest) codes which includes the appropriate access and HLC codes to route the call appropriately.

Table 91 shows examples of the construction of numbers used when dialing within the example network. Note that 6 is the Private Access code.

Table 91 Calling numbers required for DPNSS network example

Calling Site	LOC/HLC	Calling Party Number	Called Site	Dialing String	Called Party Number
Site A	244	244 1111	Site B	6 668 2222	668 2222
Site B	668	668 2222	Site D	6 848 2222	848 2222
Site D	848	2222	Site D	2229	2229
Site C	496	496 3333	Public DN	9 563 3245	563 3245

Figure 119 DPNSS networking

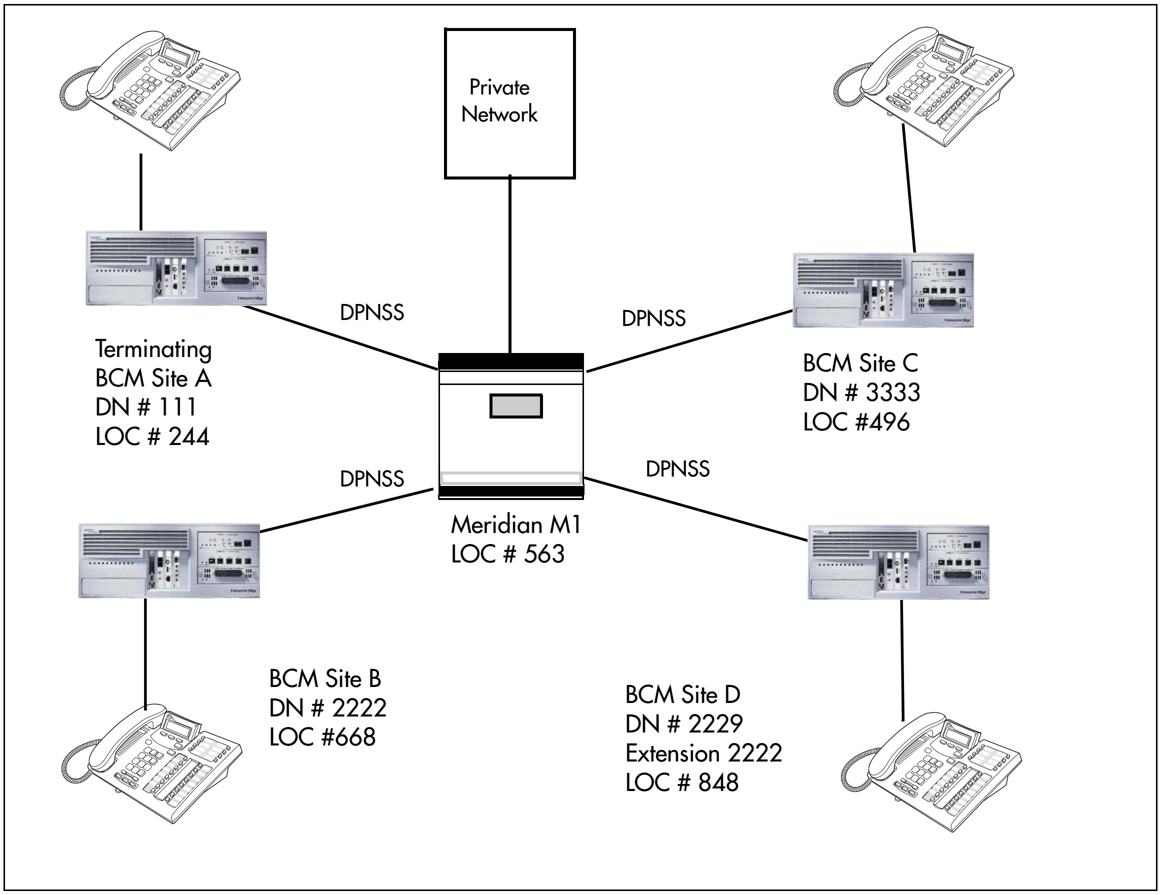

Private Network
DPNSS
DPNSS
DPNSS
DPNSS
Terminating BCM Site A DN # 111 LOC # 244
BCM Site C DN # 3333 LOC #496
Meridian M1 LOC # 563
BCM Site B DN # 2222 LOC #668
BCM Site D DN # 2229 Extension 2222 LOC # 848

Table 120 shows examples of the routing required to set up the network shown in Figure 119. Note that 6 is the Private Access code.

Figure 120 Routing for DPNSS network

Private Network: (for each branch BCM)		
Routing service to	Private network	Public network
Route	001	002
Dial out #	No number	No number
Use	Pool N	Pool N
DN type	none (private access code 6 is programmed)	public
Destination Code	6	9
Normal route	001	002
Absorb	1	1

Guidelines for creating a private dialing plan with DPNSS

Use the following guidelines when creating a private dialing plan with DPNSS.

- When creating HLCs for the nodes in your system, avoid numbering conflicts between network nodes and internal DNs, Hunt group DNs.
- Program a Private Access Code into your destination routing tables to avoid conflicts with your internal HLC and dest code dialing plan. For example, if a dialout HLC is 848, but this number already exists in the BCM system for an extension, the routing tables should add a Private Access Code to the dest code. If the code is programmed as 6, the dest code becomes 6848. 6848 uses a route to dial out 848 using the DPNSS line pool, allowing the call to be placed.

  Note that a Private Access Code is required only for specific DPNSS features such as Diversion, Route Optimization, and Redirection.

Customizing the DPNSS routing service

You can customize the routing service using the following restrictions:

- Direct Inward Access (DIA) lines allow incoming calls on private circuits to be directed to telephones without going through the normal call reception. Each DIA line is assigned to one or more extensions and is given a distinct Private Received number. When someone on another system on the network dials the Private Received number on a DPNSS line, the BCM system checks all received digits, compares the digits to an internal table and routes the call to the appropriate DIA line. All extensions programmed to have access to that DIA line will then alert for the incoming call.
- Dialing restrictions can be added to lines in line pools. Filters can restrict the use of the line to specific area codes.
- You can use host system signaling codes (“External call codes” in the *BCM 4.0 Device Configuration Guide* (N0060600)) as part of the dial out for a route. Routing can also be used as an alternate method for a direct-dial digit. For example, create a destination code 0 and program the number of the internal or external destination as the dial out. Digit absorption should be set to 1. Because overflow routing directs calls using alternate line pools, a call may be affected by different line restrictions when it is handled by overflow routing.

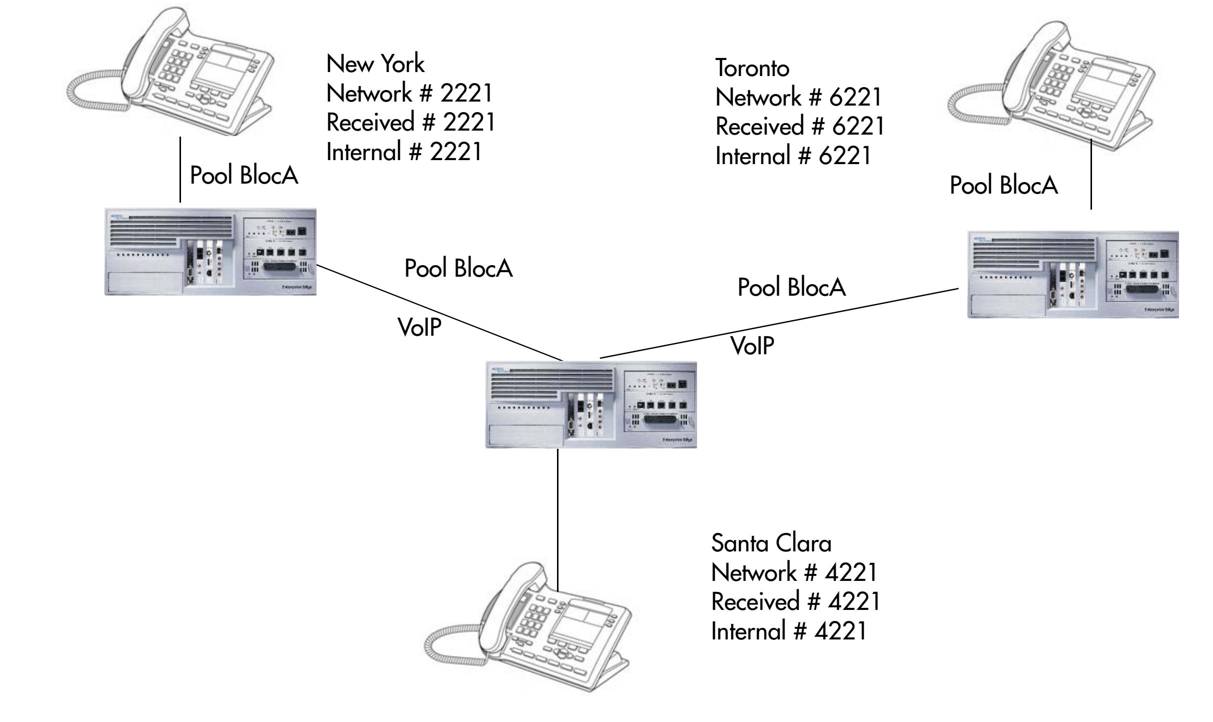
Chapter 39

Private networking: Using destination codes

By properly planning and programming routing tables and destination codes, an installer can create a dialing plan where VoIP lines between BCM are available to other systems in the network.

Figure 121 shows a network of three BCMs. Two remote systems connect to a central system.

Figure 121 Dialing plan for VoIP routing network



Each system must be running BCM software. Each system must be equipped with target lines and a VoIP keycodes with at least one IP Trunk line. Programming information for this network is shown in Table 92.

Table 92 VoIP routing for a BCM network (Sheet 1 of 3)

New York office:	
Parameter	**Setting**
Line Programming	
Network line (external)	
Line 001-004	VoIP
Line type	BlocA

Table 92 VoIP routing for a BCM network (Sheet 2 of 3)

Target line (internal)		
Line 125	Target line	
Private Received #	2221	
Line Access (set)		
Set 2221	L125: Ring only	
Line pool access	Line BlocA	
Routing service		
Route	001	
Use	BlocA	
External #	None	
Routing Destinations	**Office #1**	**Office #2**
Routing to	Santa Clara	Toronto
Destination Code	4	6
Normal route	001	001
Absorb	None	None
Dialed number:	4221	6221
Santa Clara office:		
Parameter	**Setting**	
Network line (external to New York)		
Line 001-004	VoIP	
Line type	BlocA	
Target line (internal to Santa Clara telephone)		
Line 125	Target line	
Private Received #	4221	
Line Access		
DN 4221	L125: Ring only	
Line pool access	Line BlocA	
Routing Destinations	**Office #1 and #2**	
Routing to	New York/Toronto	
Route	001	
Use	BlocA	
External #	None	
Destination Code	2	6
Absorb	None	None
Normal route	001	001

Table 92 VoIP routing for a BCM network (Sheet 3 of 3)

Remote access		
Rem access pkgs	01	
Line pool access	BlocA: ON	
Line pool access	BlocB: ON	
Toronto office:		
Parameter	**Setting**	
Trunk/Line Data (external)		
Line 001-004	VoIP	
Line type	BlocA	
Target line (internal)		
Line 125	Target line	
Private Received #	6221	
Line Access		
DN 6221	L125: Ring only	
Line pool access	Line BlocA	
Routing Destinations	**Office #1**	**Office #2**
Routing to	New York	Santa Clara
Route	001	
Use	BlocA	
External #	None	
Destination Code	4	2
Absorb	None	None
Normal route	001	001

Note: All lines in BlocA and BlocB need to be assigned in Remote Access Package 1. This is done under the restrictions tab of the lines.

If a user in New York wants to call Toronto within the network, they dial 6221. The local BCM checks the number against the routing tables and routes the call according to the destination code 6, which places the call using Route 001.

The call appears on the routing table on the BCM in Santa Clara as 6-221. Because 6 is programmed as a destination code for Toronto on the Santa Clara system, another call is placed using route 001 from Santa Clara to Toronto. At the Toronto system, the digits 6-221 are interpreted as a target line Private received number. The call now alerts at telephone 6221 in Toronto.

Note: Network calls that use routes are subject to any restriction filters in effect.
If the telephone used to make a network call has an appearance of a line used by the route, the call will move from the intercom button to the Line button.
The telephone used to make a network call must have access to the line pool used by the route.
Network calls are external calls, even though they are dialed as if they were internal calls. Only the features and capabilities available to external calls can be used.
When programming a button to dial a Network number automatically (autodial), network calls must be treated as external numbers, even though they resemble internal telephone numbers.
Routes generally define the path between your BCM and another call server in your network, not other individual telephones on that call server.

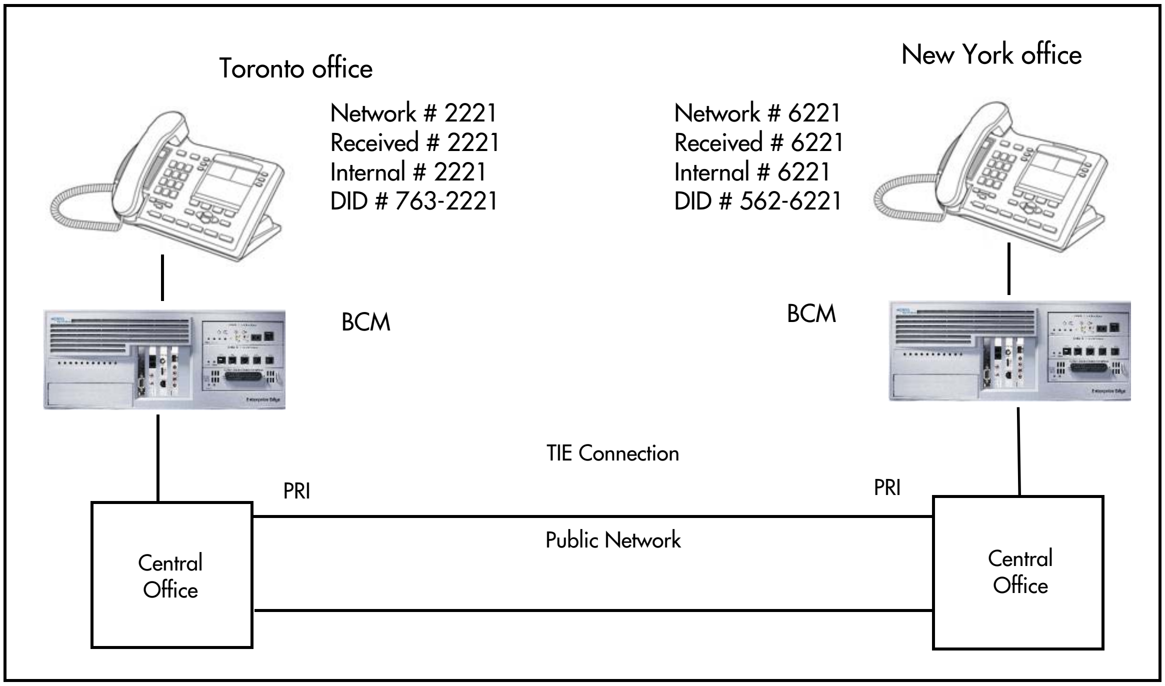
Chapter 40

Private networking: PRI Call-by-Call services

The example shown in Figure 122 highlights the use of PRI Call-by-Call services. It shows two offices of a company, one in New York and one in Toronto. Each office is equipped with a BCM and a PRI line. Each office has to handle incoming and outgoing calls to the public network. In addition, employees at each office often have to call colleagues in the other office.

Note: Call by Call Services must be provided by the Central office for them to work in the BCM.

Figure 122 PRI networking using Call-by-Call Services



To reduce long distance costs, and to allow for a coordinated dialing plan between the offices, private lines are used to handle interoffice traffic. Refer to “Dialing plan: Public network” on page 311 and “Dialing plan: Private network settings” on page 317.

If Call-by-Call services were *not* used, each BCM system might have to be equipped with the following trunks:

- 12 T1 DID lines needed to handle peak incoming call traffic
- eight T1 E&M lines needed to handle inter-office calls
- eight lines needed to handle outgoing public calls

The total required is thus 28 lines. If the BCM systems were using T1 trunks, then two T1 spans would be required at each office. Note that the total of 28 lines represents the worst case value for line usage. In reality, the total number of lines in use at any one time will generally be less than 28. For example, during periods of peak incoming call traffic, the demand for outgoing lines will be low.

With PRI Call-by-Call services, it is not necessary to configure a fixed allocation of trunks. Each of the 23 lines on the PRI can be used for DID, private TIE, or outgoing public calls. This consolidation means that it may be possible for each office to use a single PRI span, rather than two T1 spans. With PRI Call-by-Call services, the only limitation is that there are no more than 23 calls in progress at any one time.

The dialing plan at each BCM site is configured to determine the call type based on the digits dialed by the user. If a user in Toronto wishes to dial a colleague in New York, they dial the four-digit private DN (such as 6221). The dialing plan recognizes this as a private network DN, and routes the call using TIE service with a private dialing plan.

Incoming TIE calls are routed to telephones based on the digits received by the network, which in this case will be the four-digit private DN.

If a user in either location wishes to dial an external number, they dial 9, followed by the number (such as 9-555-1212). The dialing plan recognizes this as a public DN, and routes the call using Public service.

Incoming DID calls will be routed to telephones, based on the trailing portion of the digits received by the network. For example, if a public network user dials an employee in the Toronto office, the network delivers digits 4167632221. The BCM routes the call using the last four digits, 2221, to the BCM.

Refer to Table 93 for a description of the settings required for this type of routing service.

Table 93 PRI Call-by-Call services routing information (Sheet 1 of 2)

Parameter	Home System Settings
Hardware	
DTM	PRI
Protocol	NI-2
Trunk/Line Data	
Line 125	Target line
Private/Public Received #	2221
Line Access	
DN 2221	L125:Ring only
Line pool access	Line pool BlocA

Table 93 PRI Call-by-Call services routing information (Sheet 2 of 2)

Routing Services	Private Network	Public network
	New York:	**Public network**
Route	001	002
External #	No number	No number
Use	Pool BlocA	Pool BlocA
Service type	TIE	Public
ServiceID	1	N/A
DN type	Private	N/A
Destination Code	6	9
Normal route	001	002
Absorb	0	ALL
New York office:		
Parameter	**Home System Settings**	
Hardware		
DTM	PRI	
Protocol	NI-2	
Trunk/Line Data		
Line 125	Target line	
Private/Public Received #	6221	
Line Access		
DN 6221	L125:Ring only	
Line pool access	Line pool BlocA	
Routing Services	**Private Network**	**Public Network**
	Toronto	Public Network
Route	001	002
External #	No number	No number
Use	Pool BlocA	Pool BlocA
ServiceType	TIE	Public
ServiceID	1	N/A
DN type	Private	N/A
Destination Code	2	9
Normal route	001	002
Absorb	0	ALL

Chapter 41
Configuring voice messaging

You can have either an internal voice message service, or you can connect your system to an external voice message service, either over the PSTN network to a message center at the central office or through a private network to another system. This panel allows you to choose the type of voice messaging service you want to use. If you choose an external service, you can enter the contact numbers to the Centralized Voice Messaging table.

The following paths indicate where to access the loop start trunks through Element Manager and through Telset Administration:

- Element Manager: **Configuration > Applications > Voice Messaging > Contact Center**
- Telset interface: ****CONFIG > Telco features**

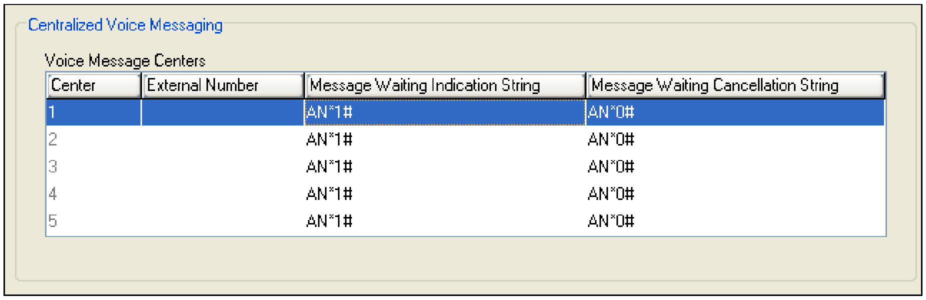
Assign external numbers to System Speed dial codes.

Panels/Subpanels	Tasks/features
"Centralized Voice Messaging (external voice mail)" on page 381	"Configuring centralized voice mail" on page 385
"Local voice messaging access (CallPilot Manager)" on page 383	Refer to the CallPilot documentation for task and feature details.
Click the navigation tree heading to access general information about Hospitality services.	

Centralized Voice Messaging (external voice mail)

This panel allows you to record on the system the dial strings that allow users on your system to access a remote voice messaging service. Note that public or private trunks need to be properly configured for these numbers to work.

Figure 123 Voice Message Centers table



Centralized Voice Messaging

Voice Message Centers

Center	External Number	Message Waiting Indication String	Message Waiting Cancellation String
1		AN*1#	AN*0#
2		AN*1#	AN*0#
3		AN*1#	AN*0#
4		AN*1#	AN*0#
5		AN*1#	AN*0#

Table 94 describes each field on this panel.

Table 94 Voice Message Centers Table

Attribute	Values	Description
Center	<read-only>	You can define a maximum of five external voice message centers. Note that any one user can only be connected to one center.
External Number	<dial string>	This is the number for the external voice message center. Ensure that you add the appropriate routing information.
Message wait indicate string (MWI)	<string>	Indicates that the message center has a message in the mailbox. This is a default NSI string for message waiting. Refer to "Programming MWI and MWC strings" on page 382.
Message wait cancellation string (MWC)	<string>	Indicates that the voice messages have been retrieved. This is a default NSI string for message waiting.

Programming MWI and MWC strings

MWI and MWC information is received from the network in the form of NSI strings.

The default MWI and MWC strings are default NSI strings for Message Waiting.
*58B*AN*1# – Message Waiting Indication
*58B*AN*0# – Message Waiting Cancellation
This provides the information required to program the strings as:
AN*1# for MWI, and
AN*0# for MWC
Private network strings will differ with different message centers. These should only be changed on the advice of your customer service representative.

DPNSS: The NSI strings in DPNSS are dependent on the supplier of the PBX. Therefore, the strings vary depending on the originating PBX system.

Each string has the following default structure: *58XYYYYY.*

Table 95 describes each part of the NSI string.

Table 95 Parts of the NSI string

String Component	Description
*58	Identifies that it is an NSI string.
X	Any letter from A to Z, or nothing.
YYYYY..	Manufacturer specific string, which can contain any sequence of alphanumeric digits or *.
#	Marks the end of the identifier.

Only the `YYYYY..` # portion of the string must be programmed for MWI and MWC. The procedure is similar to Set Name/Line Name.

The following criteria must be met when programming NSI strings for MWI/MWC:

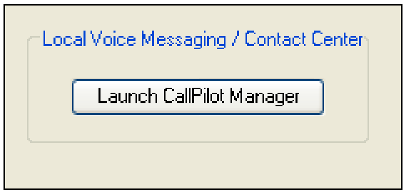
- No spaces are allowed, including spaces at the end of the string.

- A # must be present at the end.
- A # or a * cannot be present in the first character.

Local voice messaging access (CallPilot Manager)

Local voice messaging is configured using a client application. This CallPilot application is explained in detail in the CallPilot documentation.

Click the Launch CallPilot Manager button to access the application from which you can set up your local voice messaging system.



Chapter 42
Configuring centralized voice mail

The BCM supports voice mail configuration either from the local source or by accessing a remote voice mail system located on another BCM, located on a Business Communications Manager 4.0, or attached to a Meridian 1 system. The system can be configured to more than one voice mail system. However, each telephone can only be configured to one system.

Refer to the following information:

- "Local system as host" on page 385
- "Meridian system as host" on page 386
- "System set up for host system" on page 386
- "System set up for satellite systems" on page 387
- "Configuring the system for centralized voice mail" on page 389

DMS100/SL100 centralized voice mail: The BCM can also support centralized voice mail on a DMS100/SL100 switch through a PRI-DMS100 connection. The system also supports centralized voice mail on the switch through an indirect connection through an M1, where the DMS100/SL100 is connected by PRI-DMS100 to the M1 and the M1 is connected to a BCM through a PRI-MCDN connection. The DMS100/SL100 can use either the Public number or Private number of a BCM telephone to designate the mailbox number on the voice mail system.

To configure centralized voice mail, the system must be using a CDP dialing plan and be running on a private network created using either DPNSS (UK profile), PRI SL-1 or VoIP trunking set up with MCDN. Private network configuration and features are discussed in "Private networking: MCDN over PRI and VoIP" on page 329.

Note: For centralized voice mail from a DMS100/SL100 system, configure the BCM dialing plan as either CDP or UDP.

Local system as host

A local system that acts as a central voice mail location must be able to support MCDN. You can add up to 1000 mailboxes on BCM voice mail, providing you have entered adequate keycodes.

CallPilot constraints:

- To allow use of the auto attendant feature, you must ensure that the **Allow Network Transfers** check box is selected in the CallPilot Manager.
- To allow use of voice mail, you must ensure that the **Enabled Redirected DN** check box is selected in the CallPilot Manager.
- A target line must be set up to be answered by the auto attendant. The target line received digits should match the voice mail DN.

For details about setting up the CallPilot parameters and features, refer to the *CallPilot Manager Set Up and Operations Guide* and the other CallPilot supporting documentation.

Meridian system as host

If you are using a voice mail system connected to a Meridian 1 as a host system, ensure that the systems are set up to be compatible with each other.

CallPilot compatibility

If you are planning to use M-1 based CallPilot software for the voice mail system, there are no compatibility issues.

Note: CallPilot for BCM accepts network-wide and site-specific VPIM broadcast messages from M1 CallPilot, if the VPIM prefix in the message address matches the local mailbox prefix.

Meridian Mail compatibility issues

If you are using Meridian Mail as the host system, ensure the voice mail at the Meridian has the following:

- Meridian Mail rel. 7 (MM7) or above
- the appropriate number of PRI cards and D-channel handlers to support the PRI links to all the BCMs using the system.

Special requirements:

- Over a PRI SL-1 line: Meridian 1 must be on Release 19 or greater.
- Over VoIP: Meridian one must be installed with an IPT card version 3.0 or newer
- Meridian 1 requires the network ID of the BCM, select **Configuration > Telephony > Dialing Plan > Private Network** in the Element Manager. The ID is a number between 1 and 27, and is defined by the Meridian system administrator.

Also refer to “System set up for satellite systems” on page 387 for specific call features available from a Meridian 1-based voice mail system.

System set up for host system

The system that hosts the voice mail needs to ensure that incoming calls are directed to the voice mail service.

Process assumptions:

- Private network has been set up, with MCDN, between any nodes that need to access voice mail on this system.
- All systems are using the CDP dialing plan, and you have set up the correct routing to these systems.

- CallPilot or auto attendant is set up and is running for the local system.
- You have obtained a list of DNs from the remote systems that require mailboxes.

To configure the host system

1. Obtain the voice mail DN by pressing **FEATURE 985** on a system telephone.
2. If this setting matches the DN scheme for your system dialing plan, go to step 3.

   If this setting does not match the DN scheme for your system dialing plan:

   a. To access the DNs panel, select **Configuration > Telephony > Dialing Plan**.

   b. In the All DNs table, locate the DN to be changed.

   c. Double-click the number in the DN column.

   d. Enter the number obtained in step 1.
3. To access the Target Lines panel, select **Configuration > Telephony > Lines**.
4. In the Target Lines table, locate the target line to be assigned.
5. In the Details for Line subpanel, click the Assigned DNs tab.
6. Click **Add**.
7. Enter the required DN in the DN field.
8. Click **OK**.

CallPilot programming:

9. Set up CallPilot for voice mail or auto attendant answering:
   - **Voice Mail:** In CallPilot Manager click **Configuration** and **System Properties**.
     Ensure that the **Enable Redirected DN** box is selected.
   - **Auto-Attendant:** Under the **Auto-Attendant** heading, click the line record you specified in step 4 and set the Auto-Attendant to answer after 0 (zero) rings.

VoIP networking note: If you are using H.323 VoIP trunks for central voice mail, you need to set the following:

- Ensure that the local gateway protocol is set to SL-1 or CSE, based on the version of the satellite systems.
- Ensure that the remote gateways are programmed to route using CDP.
- Ensure that the remote gateway protocols are set to SL-1 or CSE, based on the version of the satellite system.

System set up for satellite systems

Systems that are remote to the voice mail system need to ensure that outgoing calls are correctly directed to the voice mail service on the host system.

Process assumptions:

- Private network has been set up, with MCDN, between the satellite and host system.
- The correct routing to the host system is set up and working.
- You have supplied a list of DNs to the host system administrator that require mailboxes.

To set up a satellite system for voice mail

1. To access the Centralized Voice Messaging panel, select **Configuration > Applications > Voice Messaging / Contact Center**.
2. Click the voice center number that you want to assign to the remote voice mail system.
3. In the External Number field, enter the voice mail DN assigned by the host system. Ensure that you include any appropriate routing codes to the string.

   Also refer to Centralized Voice Messaging (external voicemail) in the *BCM 4.0 Device Configuration Guide* (N0060600).

   DPNSS process: Type the new target number, starting with an access code, if required, or **None**. For example: **65142222**.
4. Enter the Message Waiting Indication String that is expected from the particular message center.
5. Program the Message Waiting Cancellation String that is expected from the message center.

➡ **Note:** The line must be programmed to Appear and/or Ring at the telephone.

Configuring the Target lines:

6. If the telephone does not already have a target line assigned:

   a. To access the Target Lines panel, select **Configuration > Telephony > Lines > Target Lines**.

   b. In the Target Lines table, locate the target line to be assigned.

   c. In the Details for Line subpanel, click the Assigned DNs tab.

   d. Click **Add**.

   e. Enter the required DN in the DN field.

   f. Click **OK**.

   g. Click the **Preferences** tab.

   h. In the Voice message center field, enter the center number of the voice center number that you want to assign to the remote voice mail system.
7. Repeat the previous step for all the target lines you want to change.

Configuring the telephone records:

8. To access the DNs panel, select **Configuration > Telephony > Dialing Plan**.
9. In the All DNs table, click the DN you associated with the voice mail target line.

10 In the Details for DN subpanel, click the Line Assignment tab.

11 Add the line number of the target line programmed for the telephone.

12 Select the **Vmsg** check box.

Configuring Call forward to go to voice mail:

13 For the same DN:

- **a** Click the **Capabilities and Preferences** tab.
- **b** In the Details for DN subpanel, select the **Allow redirect** check box.
- **c** Click the **Line Access** tab.
- **d** Double-click the **Fwd No Answer** field.
- **e** Enter the voice mail DN.
- **f** Double-click the **Fwd Busy** field.
- **g** Enter the voice mail DN.

14 Repeat the previous step for each of the DNs you want to assign to the remote voice mail.

15 Test the system.

VoIP networking note: If you are using H.323 VoIP trunks for central voice mail, you need to set the following:

- Ensure that the local gateway protocol is set to CSE, based on the version of the satellite systems.
- Ensure that the remote gateways are programmed to route using CDP.
- Ensure that the remote gateway protocols are set to CSE, based on the version of the satellite system.

16 Repeat for each center you want to identify.

TIPS:

- A telephone does not show that external voice messages are waiting unless you enable **VMSG set** for the lines assigned to each telephone under **Line Assignment**. Refer to “Capabilities and Preferences - Capabilities tab” in the *BCM 4.0 Device Configuration Guide* (N0060600).
- Analog telephones connected to an GASM can receive message waiting indicators if the analog line supports CLID. MWI indicator settings for analog telephones or for analog telephones attached to ATA2s, are set under the ATA heading “Configuring an analog telephone” in the *BCM 4.0 Device Configuration Guide* (N0060600).
- You can program up to five voice message center numbers, but many systems require only one.

Configuring the system for centralized voice mail

MCDN is supported over a PRI (SL-1) line or VoIP trunks between your BCM and other systems, such as Meridian 1, or Business Communications Manager systems. This section describes the specific programming for remote voice mail over PRI lines.

Apart from line configuration, MCDN over VoIP has the same system configuration.

To set up a PRI connection on the system

1 Ensure that the remote voice mail system is set up to accommodate your system on the network.

2 Ensure that your dialing plan coordinates with what the other nodes on the network are using. (select **Configuration > Telephony > Dialing Plan, Private Network panel, Private network type**)

3 Enter the network system identifier the Meridian system administrator supplied (between 1 and 27), if you are networked with a Meridian 1 somewhere in the network.
(select **Configuration > Telephony > Dialing Plan, Private Network panel, Private network type**)

4 Install a DTM module to connect to the appropriate PRI SL-1 trunk, or enter the keycode for the required number of VoIP trunks.

5 Configure the lines you plan to use, assigning them to the same line pool. Refer to "Configuring lines: PRI" on page 171 and "Configuring VoIP lines" on page 414.

6 Enter the MCDN keycode.

7 Choose the MCDN network features that you want to use.
(Select **Configuration > Telephony > Dialing Plan, Private Network** panel, and then select the MCDN subpanel)

8 Set up routing to target the PRI or VoIP line pool you set up.

9 Set up your dialing plan to recognize the network system identifiers of the other nodes on the system, so your system can pass them along, as required.

10 Assign the pool to any telephones you want to allow to use this line.

11 Program target lines and assign to telephones.

12 Set up the voice mail DN for the system that is being used as the host voice mail system for your network.

13 Test the link.

14 Refer to the CallPilot documentation to set up the mail boxes or auto attendant features and other voice mail parameters.

Chapter 43
VoIP overview

On the BCM, the LAN configuration consists of two components: Router LAN configuration, which determines how the router communicates with devices on the LAN, and Main Module LAN configuration, which determines how the Main Module of the BCM communicates with other devices on the LAN.

IP telephones

IP telephones offer the functionality of regular telephones, but do not require a hardwire connection to the BCM. Instead, they must be plugged into an IP network, which is connected to the LAN or WAN on the BCM. Calls made from IP telephones through the BCM can pass over VoIP trunks or across a Public Switched Telephone Network (PSTN).

Nortel provides two types of IP telephones. The IP telephones are wired to the IP network using Ethernet, or are accessed through your desktop or laptop computer, as in the case of the Nortel i2050 Software Phone.

VoIP trunks

VoIP trunks allow voice signals to travel across IP networks. A gateway within the BCM converts the voice signal into IP packets, which are then transmitted through the IP network to a gateway on the remote system. The device at the other end reassembles the packets into a voice signal.

Creating an IP telephony network

An IP telephony network consists of telephones, gatekeepers, IP networks, and access to a PSTN.

Networking with BCM

The BCM is a key building block in creating your communications network. It interoperates with many devices, including the Meridian 1, Succession, and other BCM devices. You can connect the BCM system to devices through multiple IP networks, as well as through the PSTN. Multiple BCM systems also can be linked together on a network of VoIP trunks and/or dedicated physical lines.

Telephones

The BCM can communicate using digital telephones (7000, 7100, 7100N, T7208, 7208, 7208N, 7316, 7316E, 7316E+KIMs, 7310), cordless telephones (7406), and IP telephones and applications (Nortel IP Phone 2001, IP Phone 2002, IP Phone 2004, Nortel i2050 Software Phone) 2007/ 11XX. With this much flexibility, the BCM can provide the type of service you require to be most productive in your business.

Note: Model 7000 phones are supported in selected markets only.

While analog and digital telephones cannot be connected to the BCM system with an IP connection, they can make and receive calls to and from other systems through VoIP trunks. Calls received through the VoIP trunks to system telephones are received through the LAN or WAN card and are translated within the BCM to voice channels.

Gatekeepers

A gatekeeper tracks IP addresses of specified devices, and provides routing and (optionally) authorization for making and accepting calls for those devices. A gatekeeper is not required as part of the network to which your BCM system is attached, but gatekeepers can be useful on networks with a large number of devices.

When planning your network, be sure to consider all requirements for a data network. Consult your network administrator for information on network setup and how the BCM fits into the network.

SIP Proxy

The SIP proxy routes calls that are not defined in the routing table. If the dialed digits do not match any of the predefined routing strings, the call is sent to the proxy IP address and then the proxy routes the call.

IP Network

WAN

A Wide Area Network (WAN) is a communications network that covers a wide geographic area, such as state or country. For BCM, a WAN is any IP network connected to a WAN card on the BCM system. This may also be a direct connection to another BCM system.

LAN

A Local Area Network (LAN) is a communications network that serves users within a confined geographical area. For BCM, a LAN is any IP network connected to a LAN card on the BCM system. Often, the LAN can include a router that forms a connection to the Internet. A BCM can have up to two LAN connections.

Key VoIP concepts

The following explains four commonly used VoIP terms.

QoS

QoS (Quality of Service) is technology that determines the quality of the VoIP connection. BCM and network routers use QoS to ensure that real time critical IP packets, such as voice packets, are given higher routing and handling priority than other types of data packets.

Silence suppression

Silence suppression also referred to as VAD (Voice activated detection) technology identifies the periods of silence in a conversation, and stops sending IP speech packets during those periods. Telco studies show that in a typical telephone conversation, only about 36% to 40% of a full-duplex conversation is active. When one person talks, the other listens. This is half-duplex. There are important periods of silence during speaker pauses between words and phrases. By applying silence suppression, average bandwidth use is reduced by the same amount. This reduction in average bandwidth requirements develops over a 20-to-30-second period as the conversation switches from one direction to another. Also refer to “Silence suppression” on page 695.

Codecs

The algorithm used to compress and decompress voice over IP networks and VoIP trunks is embedded in a software entity called a codec (COde-DECcode).

Refer to “Codec rates” on page 713 for a listing of the supported codes and their transmission rates.

- The G.711 Codec samples the voice stream at a rate of 64kbps (kilobits per second), and is the Codec to use for maximum voice quality. Choose the G.711 Codec with the companding law (alaw or ulaw) that matches your system requirements.
- The G.729 Codec samples the voice stream at 8 kbps. The voice quality is slightly lower using a G.729 but it reduces network traffic by approximately 80%.
- The G.723 Codec should be used only with third party devices that do not support G.729 or G.711.

Codecs with silence suppression, also referred to as VAD (Voice Activity Detection), make VAD active on the system, which performs the same function as having silence suppression active.Also refer to "Silence suppression" on page 695.

Note: You can only change the codec on a configured IP telephone if it is online to the BCM, or if Keep DN Alive is enabled for an offline telephone.

Proactive Voice Quality Management (PVQM)

Proactive Voice Quality Monitoring (PVQM) provides real-time notification in case of VoIP call quality degradation. This notification allows you to monitor and manage calls on the network in the real time.

PVQM monitors a set of metrics including packet loss, inter arrival jitter, round trip delay and Listening R. These metrics and supplementary information provide you with valuable insight into the real time quality of the call from the perspective of the end-user. This information can give an indication of the type of problem; this information can then be used to locate the source of the issue, thus accelerating the isolation and diagnostics phase of problem resolution.

In addition to packet loss, inter arrival jitter and round trip delay, PVQM monitors the "Listening R" value. The R-Factor, as defined by ITU G.107 and IETF 3611, is a call quality index that assesses network impairments such as packet drops, jitter and round trip delay with consideration for the burstiness and recency of these impairments. The Listening R metric provides you with definitive answers about the actual QoS delivered to the telephone user. With this metric, you can see the raw data (such as jitter or packet drop rate), and a summary of the effect of the data on the quality experienced by the user.

For example, a Warning Threshold for the listening R-value might be set at 80. When voice quality drops below this value as measured at the telephone set itself, an event is generated. The event notification is augmented with other state information, such as network loss rate, average rate of discards due to jitter, average length of bursts, and presented as an alarm. Analysis of the alarms and supplementary information in the alarm description helps you identify and troubleshoot voice quality issues and proactively initiate responsive actions.

Refer to the *BCM 4.0 Administration Guide* (N0060598) for information on how to configure and use PVQM functionality.

Chapter 44
VoIP trunk gateways

With a VoIP trunk, you can establish communications between a BCM and a remote system across an IP network. Each trunk is associated with a line record (lines 001-060), and are configured in the same way that other lines are configured.

However, VoIP trunks have additional programming to support the IP network connection.

This system supports SIP and H.323 trunk protocols. Both types of trunks support connections to other BCMs, a central call server such as Succession 1000/M, and trunk-based applications. SIP trunks and H.323 trunks are assigned to a single Pool, and the routing decision to route calls via H.323 or SIP is made based on the routing modes of the two services (Direct/Gatekeeper/Proxy) and the combined routing table.

The following path indicates where to access the Voice over IP (VoIP) trunk gateway in Element Manager:

- Element Manager: **Configuration > Telephony Resources > IP Trunks > > H323 Settings/ SIP Settings** tabs

Configuring a VoIP trunk requires the following:

- "Pre-installation system requirements" on page 395
- "Keycodes" on page 396
- "H.323 network applications considerations" on page 396
- "SIP network applications considerations" on page 396

VoIP trunks can be used for calls originating from any type of telephone within the BCM system. Calls coming into the system over VoIP trunks from other systems can be directed to any type of telephone within the system.

You cannot program Auto DN or DISA DN for VoIP trunks; therefore, you cannot use CoS passwords to remotely access features on your system. The exception to this would be a tandemned call, where a call comes into system A over the PSTN, then tandems to system B over a VoIP trunk. In this case, the remote access package on the line will determine which system features are available to the caller.

Pre-installation system requirements

Ensure that you have obtained the following information or familiarize yourself with the requirements before continuing with VoIP trunk configuration:

Keycodes

Before you can use VoIP, you must obtain and install the necessary keycodes. See the *Keycode Installation Guide* (N0060625) for more information about installing the keycodes. Talk to your BCM sales agent if you need to purchase VoIP keycodes.

Each keycode adds a specific number of VoIP trunks. You must reboot your BCM after you enter VoIP keycodes to activate trunking.

If you want to use the MCDN features on the VoIP trunks, you will need an MCDN keycode. If you have already deployed MCDN for your SL-1 PRI lines, you do not require an additional keycode.

H.323 network applications considerations

In order to ensure a level of quality during call setup, QoS monitor must be enabled and configured.

If your network uses a gatekeeper (H.323 trunks only), there are also specific settings that must be set on the Local Gateway panel to recognize the gatekeeper, and also within the gatekeeper application, so that VoIP lines are recognized. Also refer to gatekeeper configuration “VoIP interoperability: Gatekeeper configuration” on page 417.

If you plan to use H.323 trunking and you have a firewall set up, ensure that the ports you intend to use have been allowed.

SIP network applications considerations

In order to maintain a level of quality during call setup, QoS monitor must be enabled and configured.

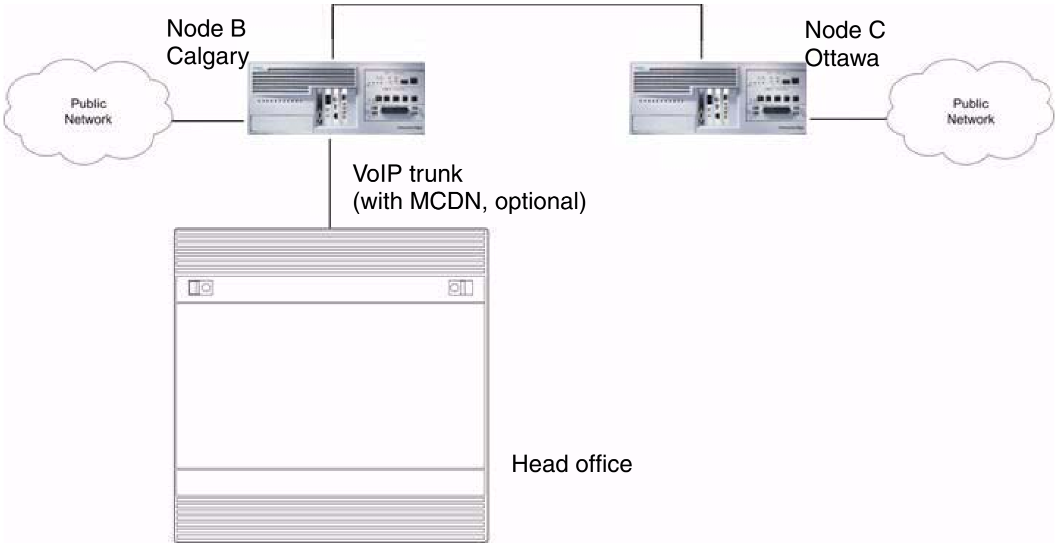
SIP URI maps of both endpoints must match.

If you plan to use SIP trunking and you have a firewall set up, ensure that the ports you intend to use have been allowed.

How VoIP trunks make a network

Figure 124 shows a simple private networking configuration of three systems connected by VoIP trunks. As in all private networking, each system has direct routing configurations to the directly adjacent systems. As well, the dialing plans are configured to ensure that remote calls are correctly routed to the receiving system, such as, if Node A called someone in Node C.

Figure 124 Internal call from Meridian 1 tandems to remote PSTN line



Since the VoIP trunks are configured into line pools, you can assign line pool codes to users who have been assigned access to the VoIP trunks. However, if you intend to set up your system to use fallback, so that calls can go out over PSTN if the VoIP trunks are not available, you must use routes and destination codes to access the VoIP trunk line pools.

Local gateway programming

The VoIP trunk access point at each system is called a gateway. The gateway to your system, the local gateway, determines how incoming and outgoing calls will be handled.

The H323 and SIP Media Parameters tabs determine a number of system settings. These values need to be coordinated with the other systems on the network to ensure that all features work consistently across the network. Media parameters include setting:

- the order of preferred codecs
- voice activity detection
- jitter buffer size
- codec payload size
- IP fax transmission availability on the network

The local gateway parameters define how the BCM prefers call signaling information to be directed through VoIP trunks. Call signaling establishes and disconnects a call.

If the network has a gatekeeper (H.323 trunks only), the BCM can request a method for call signaling, but whether this request is granted depends on the configuration of the gatekeeper. Ultimately, the gatekeeper decides which call signaling method to use.

H323 Settings include:

- fallback to circuit switched availability

- type of call signaling, either directly to the far end system or through a network gatekeeper
- if there is a gatekeeper, the relevant IP information is noted
- a KeepAlive signal timer
- the protocol the system will use for the gateway (must be compatible with remote system or gatekeeper)
- allowing/disallowing VoIP gateway tunnel H.245 messages within H.225
- being able to identify unique call signaling and RAS ports

Notes about NPI-TON aliases for H.323 trunks

NPI-TON aliases store dialed number prefixes as well as information about the type of number. A dialed number can be qualified according to its Type of Number (TON), as well as its Numbering Plan Identification (NPI). Nortel recommends this format over the E.164 format, for encoding dialed numbers and aliases registered with a gatekeeper.

When using a gatekeeper, and attempting to place an outgoing VoIP trunk call, ensure that the route and dialing plan configuration matches the NPI-TON aliases registered, by the destination, with the gatekeeper. These requirements are summarized in Table 96.

Table 96 Route and Dialing Plan configurations for NPI-TON

Route (DN type)	**Dialing Plan used by calling gateway**	**Alias configured for calling gateway (“alias name” in Element Manager)**
Public	Public	PUB:<dialedDigitsPrefix>
Private	Private (Type = None)	PRI:<dialedDigitsPrefix>
	Private (Type = CDP)	CDP:<dialedDigitsPrefix>
	Private (Type = UDP)	UDP:<dialedDigitsPrefix>

Routing Table

Since VoIP trunks are point-to-point channels, besides the local gateway information on your system, you need to tell your system about the gateway at the remote end.

However, if the network has a gatekeeper or a SIP Proxy Server, it handles call traffic, so a routing table is not required.

To configure a remote gateway, you need to define the following information:

- a name that identifies the destination system
- the IP address of the destination system
- whether QoS monitor is enabled (this is required if you plan to use PSTN fallback)
- transmit threshold so that the system knows when to activate the fallback feature
- the remote gateway system type
- the gateway protocol

- the unique digit(s) that identify the remote system. This is usually part of the destination code.

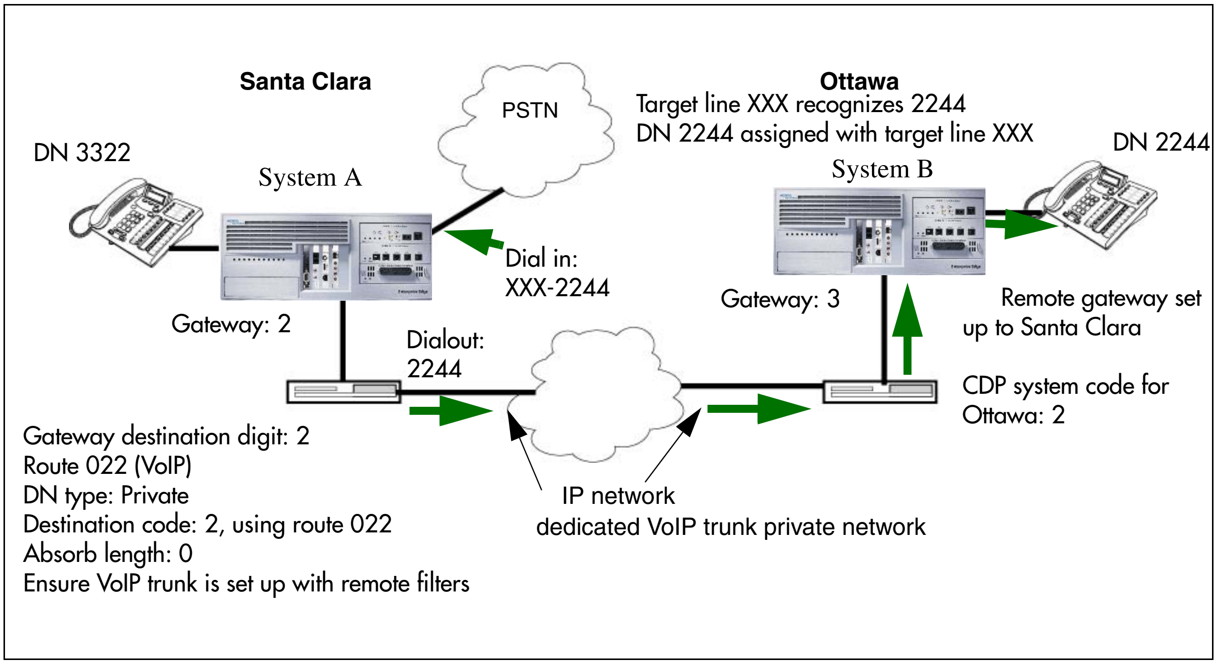
PSTN call to remote node

Making a call to a remote node requires any BCM systems between the calling and receiving nodes to have the correct routing to pass the call on to the next node. This is the same if you use PSTN lines or VoIP trunks for the network.

Figure 125 shows a call tandeming from the public network (PSTN), through System A (Santa Clara) and being passed to System B (Ottawa) over a VoIP trunk network. In this case, it might be a home-based employee who wants to call someone in Ottawa.

You cannot program DISA for VoIP trunks, therefore, your system cannot be accessed from an external location over a VoIP trunk. The exception to this is if the call comes into a tandemed system (system A) from a PSTN, and the call is then sent out across a VoIP trunk to system B, as in this example. In this case, system A is controlling remote access through remote access packages and routing, transferring the outside call to a VoIP trunk, which is accessed by an allowed dial sequence. The VoIP trunk connects directly to system B, where the dialing sequence is recognized as directed to an internal DN. In this scenario, all remote call features are available to the caller.

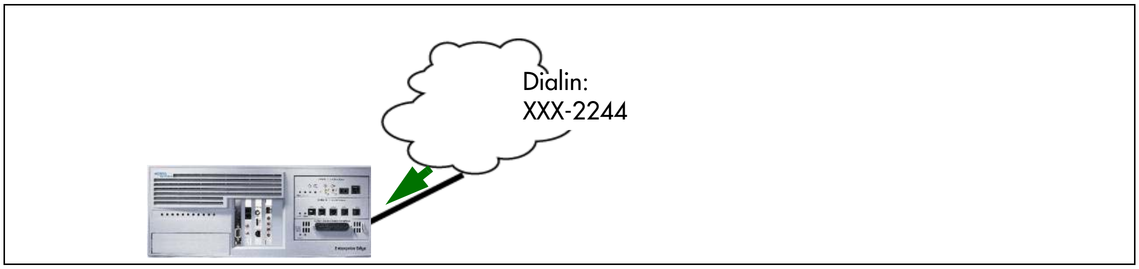
Figure 125 Calling into a remote node from a public location


Santa Clara
PSTN
Ottawa
Target line XXX recognizes 2244
DN 2244 assigned with target line XXX
DN 3322
System A
System B
DN 2244
Dial in:
XXX-2244
Gateway: 3
Remote gateway set up to Santa Clara
Gateway: 2
Dialout:
2244
CDP system code for Ottawa: 2
Gateway destination digit: 2
Route 022 (VoIP)
DN type: Private
Destination code: 2, using route 022
Absorb length: 0
Ensure VoIP trunk is set up with remote filters
IP network
dedicated VoIP trunk private network

Call process

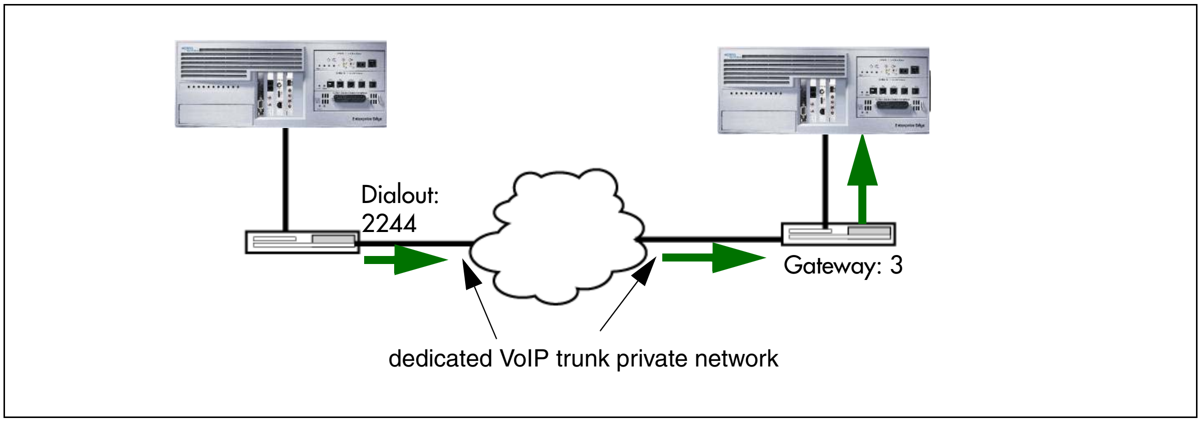
Based on Figure 125, this is how the call would progress:

1 A home-based employee in Santa Clara wants to call someone in Ottawa, so they dial into the local BCM network using the access code for an unsupervised trunk (not VoIP trunks) and the destination code and DN for the person they want to reach on System B.

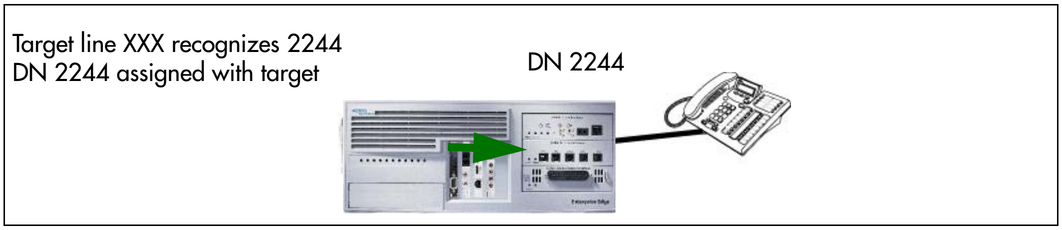


2 When the call is received from the public network at System A (Santa Clara), the system recognizes that the received number is not a local system number. The call is received as a public call.

3 System A has a route and destination code that recognizes the received number and destination code as belonging to the route that goes to System B (Ottawa). System A passes the call to System B over a dedicated trunk, in this case, a VoIP trunk. This call is now designated as a private call type.



4 System B recognizes the code as its own, and uses a local target line to route the call to the correct telephone.



Fallback to PSTN from VoIP trunks

Fallback is a feature that allows a call to progress when a VoIP trunk is unavailable or is not providing adequate quality of service (QoS).

Refer to the information under "Describing a fallback network" on page 401 for details about setting up fallback for VoIP trunks.

By enabling **PSTN fallback** on the Local Gateway IP Interface panels for SIP and H.323 trunks, you allow the system to check the availability of a VoIP trunk, then switch the call to a PSTN line. For the PSTN fallback to work on a suitable bandwidth, QoS monitor must be enabled and a transmit threshold must be set. For QoS and transmit threshold settings refer to Table 98.

You use scheduling and destination codes to allow the call to switch from SIP or H.323 to a PSTN line without requiring intervention by the user.

Use the dialing plan worksheet in the Programming Records to plan your dialing requirements so you can pinpoint any dialing issues before you start programming. If you are programming an existing system, you can look at what numbers the users are familiar with dialing, and you can attempt to accommodate this familiarity into your destination codes plan.

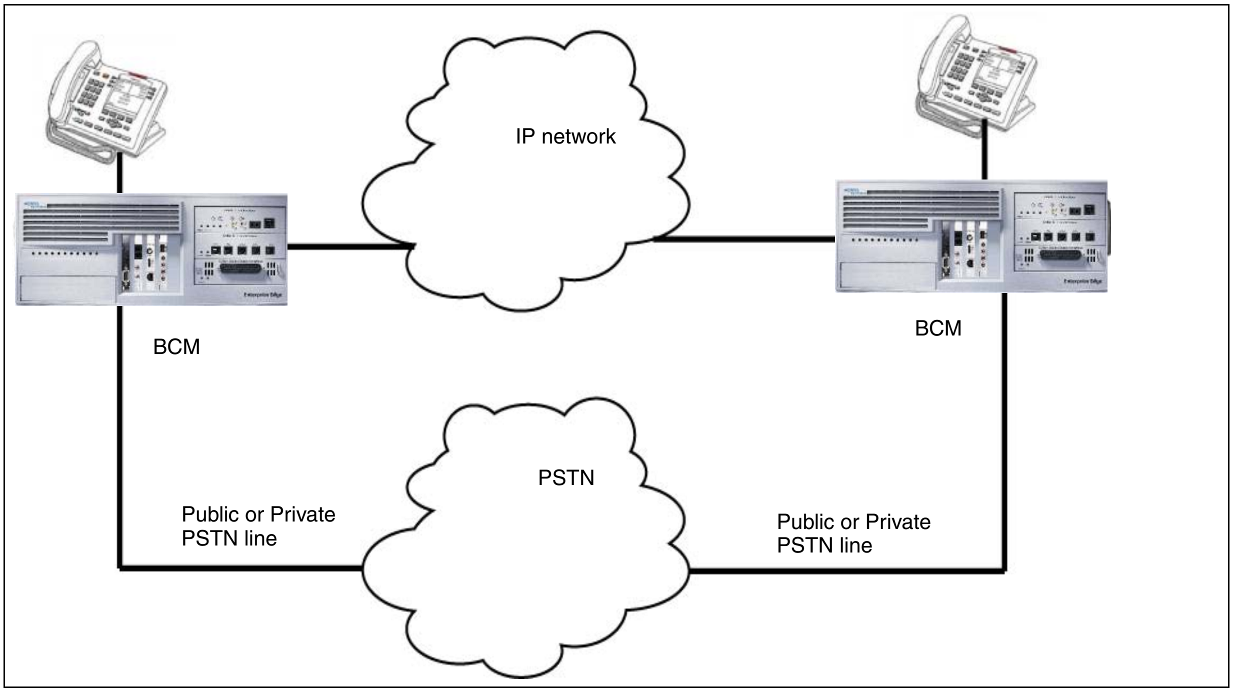
On any IP gateway for which you want to allow fallback based on network quality, you need to ensure that QoS monitor is enabled.

Warning: QoS monitor must be turned on at both endpoints. To enable the QoS Monitor select **Configuration > Telephony Resources > IP Trunks > Routing Table**.

Describing a fallback network

Figure 126 shows how a fallback network would be set up between two sites.

Figure 126 PSTN fallback diagram



In a network configured for PSTN fallback, there are two connections between a BCM and a remote system.

- One connection is a VoIP trunk connection through the IP network.
- The fallback line is a PSTN line, which can be the public lines or a dedicated T1, BRI, PRI or analog line, to the other system.

When a user dials the destination code, the system checks first to see if the connection between the two systems can support an appropriate level of QoS (if enabled). If it can, the call proceeds as normal over the VoIP trunk. If the minimum acceptable level of QoS is not met, the call is routed over the second route, through the PSTN line.

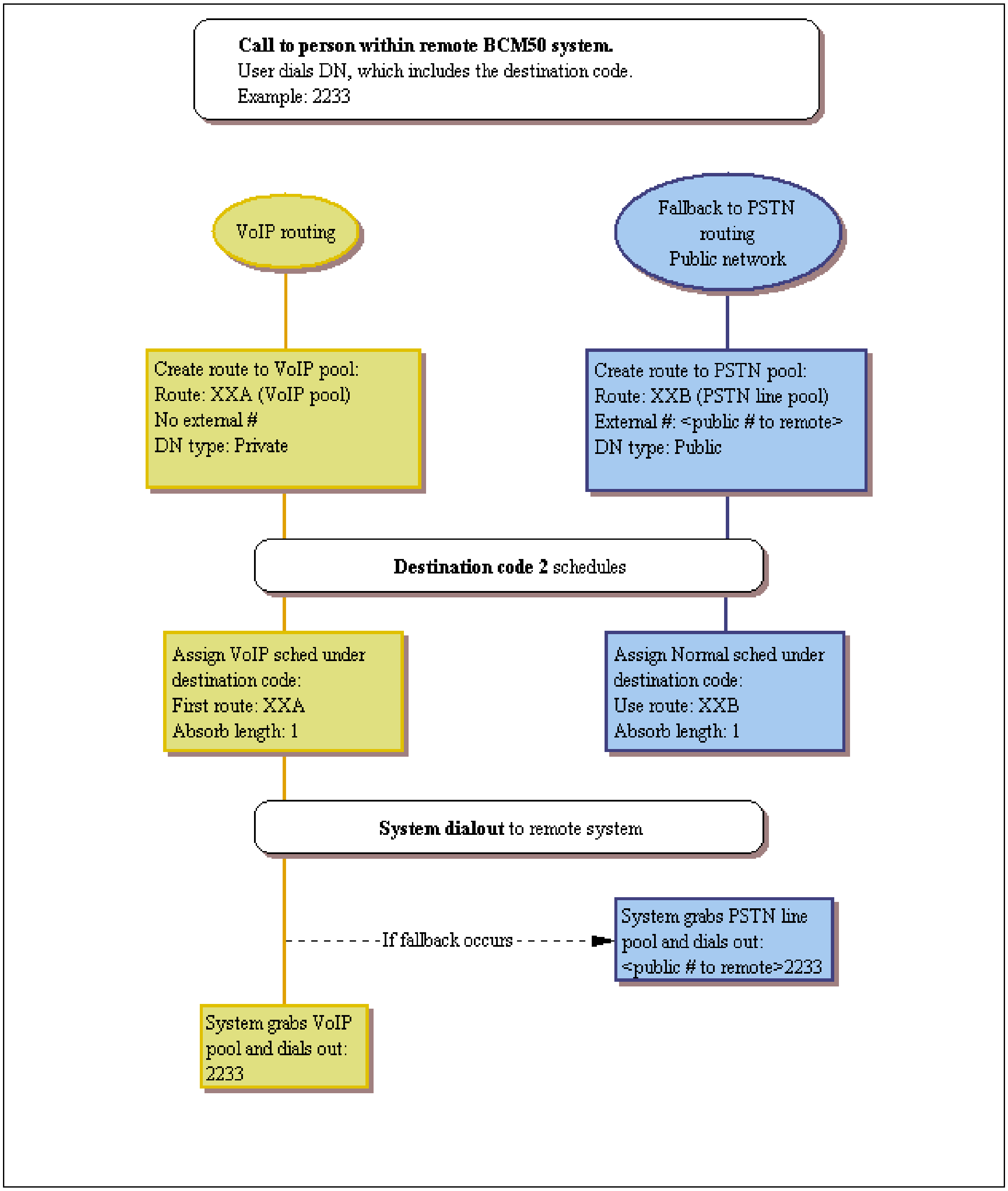
In many cases, this involves configuring the system to add and/or absorb digits.

For detailed information about inserting and absorbing digits, see "Dialing plans" on page 247.

How fallback routing works

CDP network: User dials 2233 (remote system DN: 2233; remote identifier/destination digit: 2). The system absorbs the 8, no other digits are absorbed and the system dials out 2233.
If the call falls back to PSTN line, the system still only absorbs the 8. If the PSTN line is on a private network, the system dials out 2233. If the PSTN line is a public line, the system dials out the public access number to the remote system in front of the 2233. Refer to Figure 127.

Figure 127 Setting up routes and fallback for call to remote system (CDP dialing code)


Call to person within remote BCM50 system.
User dials DN, which includes the destination code.
Example: 2233
VoIP routing
Fallback to PSTN routing Public network
Create route to VoIP pool:
Route: XXA (VoIP pool)
No external #
DN type: Private
Create route to PSTN pool:
Route: XXB (PSTN line pool)
External #: <public # to remote>
DN type: Public
Destination code 2 schedules
Assign VoIP sched under destination code:
First route: XXA
Absorb length: 1
Assign Normal sched under destination code:
Use route: XXB
Absorb length: 1
System dialout to remote system
If fallback occurs
System grabs PSTN line pool and dials out:
<public # to remote>2233
System grabs VoIP pool and dials out:
2233

UDP network: The user dials 2233 (remote system DN: 2233; destination digits/private access code: 555). The system then adds the private access code to the dialout digits.
If the call falls back to PSTN line, the system then dials out the private access code (private network PSTN line) or public access number (public PSTN) to the remote system in front of the 2233.

Optional VoIP trunk configurations

There are a number of VoIP trunk features that are optional to setting of VoIP trunk functions. The following section briefly describes these features:

- Port settings (firewall): In some installations, you may need to adjust the port settings before the BCM can work with other devices.

  Firewalls can interfere with communications between the BCM and another device. The port settings must be properly configured for VoIP communications to function properly. Using the instructions provided with your firewall, ensure that communications using the ports specified for VoIP are allowed.

  A Nortel IP telephone uses ports between 51000 and 51200 to communicate with the system. The system, by default, uses ports 28000 to 28255 to transmit VoIP packets.

  BCM uses UDP port ranges to provide high priority to VoIP packets in existing legacy IP networks. You must reserve these same port ranges and set them to high priority on all routers that an administrator expects to have QoS support. You do not need to reserve port ranges on DiffServ networks.

  You can select any port ranges that are not used by well-known protocols or applications.

  Each H.323 or VoIP Realtime Transfer Protocol (RTP) flow uses two ports, one for each direction. The total number of UDP port numbers to be reserved depends on how many concurrent RTP flows are expected to cross a router interface. In general:

  - Include port number UDP 5000 in the reserved port ranges, for the QoS monitor.
  - The port ranges reserved in a BCM system are also reserved by the remote router.
  - You must reserve two ports for each voice call you expect to carry over the IP network.
  - You can reserve multiple discontinuous ranges. BCM requires that each range meet the following conditions: Each range must start with an even number; each range must end with an odd number; no more than 256 ports can be reserved.
- Gatekeepers: The BCM supports the use of an ITU-H323 gatekeeper. A gatekeeper is a third-party software application residing somewhere on the network, which provides services such as:
  - address translation
  - call control
  - admission control
  - bandwidth control
  - zone management
  - IP registration

A single gatekeeper manages a set of H.323 endpoints. This unit is called a Gatekeeper Zone. A zone is a logical relation that can unite components from different networks (LANS). These Gateway zones, such as the BCM, are configured with one or more alias names that are registered with the gatekeeper. The gatekeeper stores the alias-IP mapping internally and uses them to provide aliases to IP address translation services. Later, if an endpoint IP address changes, that endpoint must re-register with the gatekeeper. The endpoint must also re-register with the gatekeeper during the time to live (TTL) period, if one is specified by the gatekeeper.

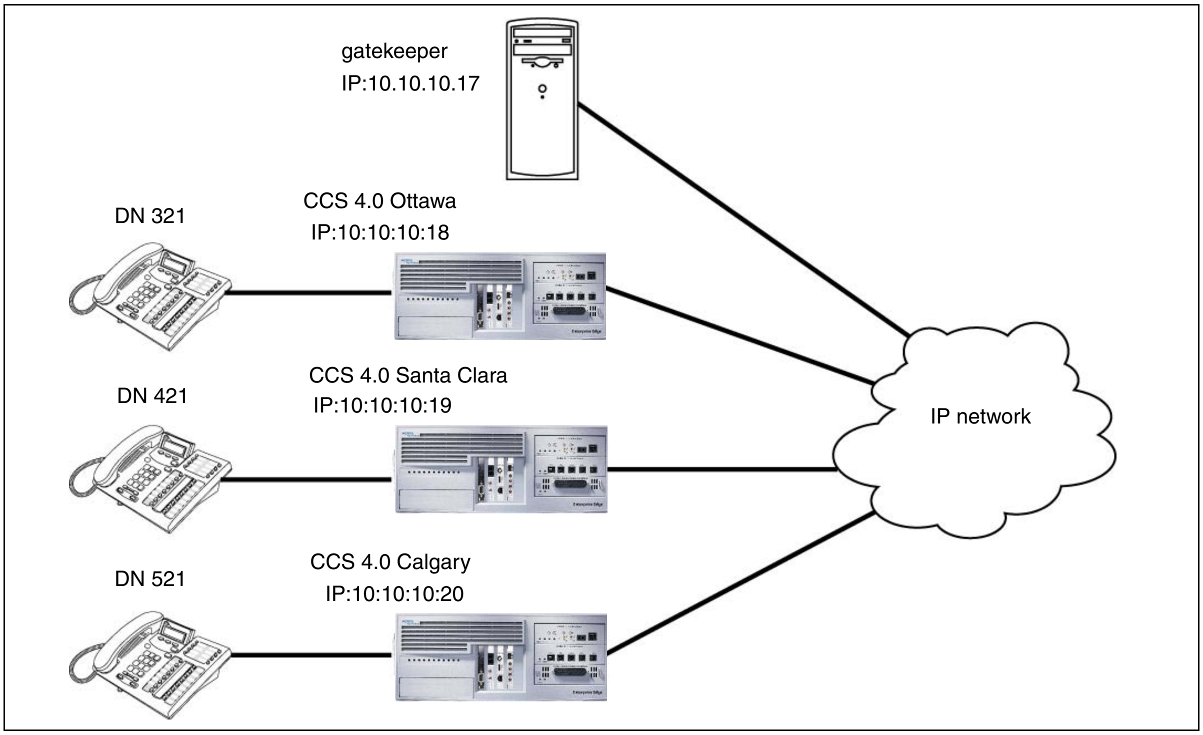
Refer to the gatekeeper software documentation for information about changing IP addresses.

Note: A gatekeeper may help to simplify IP configuration or the BCM dialing plan; however, it does not simplify the network dialing plan.

Gatekeeper call scenarios

This section explains how a call would be processed for the two types of gatekeeper configurations. Figure 128 shows a network with three BCMs and a gatekeeper.

Figure 128 BCM systems with a gatekeeper



This example explains how a call from DN 321 in Ottawa would be made to DN 421 in Santa Clara. It assumes that call signaling is set to Gatekeeper Resolved and no pre-granted AdmissionRequest (ARQ) has been issued:

1. BCM Ottawa sends an ARQ to the gatekeeper for DN 421.
2. The gatekeeper resolves DN 421 to 10.10.10.19 and returns this IP in an AdmissionConfirm to the BCM Ottawa.

3. BCM Ottawa sends the call Setup message for DN 421 to the gateway at 10.10.10.19, and the call is established.

If call signaling is set to Gatekeeper Routed and no pre-granted ARQ has been issued:

1. BCM Ottawa sends an ARQ to the gatekeeper for DN 421.
2. The gatekeeper resolves DN 421 to 10.10.10.17.
3. BCM Ottawa sends the call Setup message for DN 421 to the gatekeeper (10.10.10.17), which forwards it to the gateway at 10.10.10.19.
4. The call is established.

- Faxing over VoIP trunks: You can assign VoIP trunks to wired fax machines if you have T.38 fax enabled on the local gateway. The BCM supports this IP fax feature between BCMs, BCM200/400/1000 running BCM 3.5 and subsequent up-level versions of software, and a Meridian 1 running IPT 3.0 (or newer) software, or a CS 1000/M.

  The system processes fax signals by initiating a voice call over the VoIP line. When the T.38 fax packets are received at the remote gateway, the receiving system establishes a new path that uses the T.38 protocol. Both the endpoints must be running a software version that supports the T.38 fax.

Caution: Operations note: Fax tones that broadcast through a telephone speaker may disrupt calls at other telephones using VoIP trunks in the vicinity of the fax machine. Here are some suggestions to minimize the possibility of your VoIP calls being dropped because of fax tone interference:

- Locate fax machine away from other telephones.
- Turn the speaker volume on the fax machine to the lowest level, or off.

Fax tones recorded in a voice mail box: In the rare event that fax tones are captured in a voice mail message, opening that message from an telephone using a VoIP trunk will cause the connection to fail.

For a list of limitations and requirements for using T.38 fax, refer to "Operational notes and restrictions" on page 406.

Operational notes and restrictions

Some fax machines will be unable to successfully send faxes over VoIP (T.38) trunks to the following destinations:

- CallPilot mailboxes
- CallPilot mailboxes (accessed through auto-attendant)
- Fax Transfer (calls transferred to a system fax device through the auto-attendant)
- Use the following tips to avoid this problem:
  - Avoid the use of manual dial on the originating fax machine. In some fax machines, manually dialing introduces a much shorter call time-out.

- If manual dial must be used, then the user should wait until the call is answered before starting the fax session.
- If manual dial must be used, then the user should enter the digit **8** before initiating the fax session. This ensures that the fax session is initiated by CallPilot before the fax machine's timer is started.
- The call duration can be increased by adding a timed pause to the end of dialing string (for example: 758-5428,,,,). This allows the call to ring at the destination before the fax machine call duration timer starts.
- Since the problem is related to the delay in initiating the fax session, the number of rings for fax mailboxes Call Forward No Answer (CFNA) should be minimized.

Table 97 is a list of restrictions and requirements for the T.38 fax protocol.

Table 97 T.38 restrictions and requirements

Supported	Not supported
only UDP transport	TCP
only UDP redundancy	Forward Error Correction (FEC)
T.38 version 0	Fill removal
on H.323 VoIP trunks between BCMs, between BCMs and legacy BCMs, or between BCM and Meridian 1-IPT and CS 1000/M	MMR transcoding JBIG transcoding

Chapter 45

Configuring VoIP trunk gateways

The following explains how to configure voice over IP (VoIP) trunks on a BCM system for incoming traffic. A VoIP trunk allows you to establish communications between a BCM and a remote system across an IP network.

The following path indicates where to where to configure VoIP trunks in Element Manager:

- Element Manager: **Configuration > Resources > Telephony Resources > IP Trunks**

Task: Set up VoIP gateway parameters

- Set up the media parameters for the gateway. ("Configuring VoIP trunk media parameters" on page 410)
- Set up the local gateway parameters, including H323 gatekeeper or SIP Proxy settings, if necessary ("Setting up the local gateway" on page 410)
- Set up the routing table, if one is required. ("Setting up remote gateways" on page 412)
- Configure the line parameters. ("Configuring VoIP lines" on page 414)

Prerequisites

Ensure that you have obtained the following information or familiarize yourself with the requirements before continuing with VoIP trunk configuration:

- Keycodes: Obtain and install the necessary keycodes for the number of VoIP trunks you want to support on the system. See the *Keycode Installation Guide* (N0060625) for more information about installing the keycodes. Talk to your BCM sales agent if you need to purchase VoIP keycodes.

  Each keycode adds a specific number of VoIP trunks. You must reboot your BCM after you enter VoIP keycodes to activate trunking.

  The FEPS service will restart automatically after you enter the VoIP keycodes.

  If you want to use the MCDN features on the VoIP trunks, you will need an MCDN keycode. If you have already deployed MCDN for your SL-1 PRI lines, you do not require an additional keycode.
- Media gateway parameters: Ensure that the gateway parameters are set correctly for the IP trunks.
- H.323 network applications considerations:

  - — If your network uses a gatekeeper (H.323 trunks only), there are also specific settings that must be set on the your system to recognize the gatekeeper, and also within the gatekeeper application, so that VoIP lines are recognized. Refer to "VoIP interoperability: Gatekeeper configuration" on page 417. If there is a gatekeeper on the network, you do not have to configure remote gateway settings.
  - — If you plan to use H.323 trunking, and you have a firewall set up, ensure that the ports you intend to use have been allowed.
- SIP network applications consideration:
  - — If you plan to use SIP trunking, and you have a firewall set up, ensure that the ports you intend to use have been allowed.

"Using VoIP to tandem systems" on page 361, and "Configuring fallback over a VoIP MCDN network" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Configuring VoIP trunk media parameters

The VoIP trunk media parameters allow you to specify the order in which the trunk will select IP telephony system controls for codecs, jitter buffers, voice activity detection and payload size. There are two types of routing protocols used for VoIP trunks H323 and SIP. Refer to the following procedures to configure the settings for these protocols.

To configure H323 media parameters

1. Click **Configuration > Resources > Telephony Resources > IP trunks**
2. Select the **H323 Media Parameters** tab in the bottom panel.
3. Enter the information that supports your system. Refer to the information in Table 35. Ensure that these settings are consistent with the other systems on the network.
4. Set up the local gateway parameters. ("Setting up the local gateway" on page 410)

To configure SIP media parameters

1. Click **Configuration > Resources > Telephony Resources > IP trunks**
2. Select the **SIP Media Parameters** tab in the bottom panel.
3. Enter the information that supports your system. Refer to the information in Table 37. Ensure that these settings are consistent with the other systems on the network.
4. Set up the local gateway parameters. ("Setting up the local gateway" on page 410)

Setting up the local gateway

The call signaling method used by the local gateway defines how the BCM prefers call signaling information to be directed through VoIP trunks. Call signaling establishes and disconnects a call. You set this information in the local gateway panels.

If the network has a gatekeeper (H.323 trunks, only), the BCM can request a method for call signaling, this request is granted depending on the configuration of the gatekeeper. Ultimately, the gatekeeper decides which call signaling method to use. Refer to "VoIP interoperability: Gatekeeper configuration" on page 417.

The following path indicates where to access the local gateway in Element Manager:

Element Manager: **Configuration > Resources > Telephony Resources > IP Trunks**

1 On the Modules panel, in the Module type column, select the IP Trunks line.

2 In the bottom panel, select the Local Gateway tab.

3 Choose the settings that you need for your system:

- Fallback to circuit-switched: define how you want the system to handle calls that the system fails to send over the VoIP trunk.

➡ **Note:** Enabled-TDM enables fallback for calls originating on digital telephones. This is useful if your IP telephones are connected remotely, on the public side of the BCM network, because PSTN fallback is unlikely to result in better quality of service in that scenario.

- Forward redirected OLI - If the box is selected, the OLI of an internal telephone is forwarded over the VoIP trunk when a call is transferred to an external number over the private VoIP network. If the box is cleared, only the CLID of the transferred call is forwarded.
- Send name display - When selected, the telephone name is sent with outgoing calls to the network.
- Remote capability MWI - This setting must coordinate with the functionality of the remote system hosting the remote voice mail.
- Call Signaling: Determine how the calls are delivered over the network:
  - — **Direct**: call signaling information is passed directly between endpoints.
    Note: You will need to set up remote gateways ("Setting up remote gateways" on page 412).
  - — **Gatekeeper Resolved**: all call signaling occurs directly between H.323 endpoints. This means that the gatekeeper resolves the phone numbers into IP addresses, but the gatekeeper is not involved in call signaling.
  - — **Gatekeeper Routed**: uses a gatekeeper for call setup and control. In this method, call signaling is directed through the gatekeeper.
  - — **Gatekeeper Routed no RAS:** Use this setting for a NetCentrex gatekeeper. With this setting, the system routes all calls through the gatekeeper but does not use any of the gatekeeper Registration and Admission Services (RAS).
  - — Refer to "Using CS 1000 as a gatekeeper" on page 417 for specific information about configuring the gatekeeper for H.323 trunks.
    Network note: If your private network contains a Meridian 1-IPT, you cannot use Radvision for a gatekeeper.
- Call signaling port: If there are VoIP applications that require non-standard call signaling ports, enter the port number here. 0 = the system uses the first available port.

- RAS port: If the VoIP application requires a non-standard RAS port, enter the port number here. 0 = the system uses the first available port.
- Enable H245 tunneling: Select or deselect the check box to allow or disallow H.245 messages within H.225. Note that the VoIP Gateway service must be restarted for any change to take effect.
- Gatekeeper Support: Fill out these fields if the network is controlled by a Gatekeeper: Also refer to "VoIP interoperability: Gatekeeper configuration" on page 417.
  - — Primary Gatekeeper IP: This is the IP address of the primary gatekeeper.
  - — Backup Gatekeepers: NetCentrex gatekeeper does not support RAS, therefore, any backup gatekeepers must be entered in this field. Gatekeepers that use RAS can provide a list of backup gatekeepers for the end point to use in the event of the primary gatekeeper failure.
- In the Alias names field, enter all the alias names required to direct call signals to your system.
- Gateway protocol - Select SL1 for BCM 2.5 systems. Select CSE for BCM 3.0 and newer systems. Or select None.
- Registration TTLs: Specifies the KeepAlive interval
- Gateway TTLs: This protocol should match all other systems on the network.
- Status: This field displays the current status of the gatekeeper.

4 Suggested next steps:

- Ensure router settings, firewalls and system ports are set correctly to support IP traffic over the trunks.
- "Configuring lines" on page 147
- "Configuring lines: Target lines" on page 177
- "Setting up VoIP trunks for fallback" on page 423
- Ensure private network dialing plan and access settings matches the rest of the private network: "Dialing plan: Private network settings" on page 317
- Private networking: "Private networking: Basic parameters" on page 349
- Assigning the VoIP line pools to system telephones: "Line Access - Line Pool Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600).

Setting up remote gateways

The following explains how to set up your system to place calls through VoIP trunks. The system at the other end of the call must be set up to receive VoIP calls. For information about this, refer to "Configuring a remote gateway (H.323 trunks)" on page 413.

Configuration note: If the VoIP network has a gatekeeper, you do not need to configure remote gateways, as they are not used.

Configuring a remote gateway (H.323 trunks)

The following explains how to configure the BCM to communicate with other BCMs and/or other VoIP gateways such as Meridian IPT using H.323 trunks. The remote gateway list must contain an entry for every remote system to which you want to make VoIP call.

Gatekeeper note: If your system is controlled by a gatekeeper, you do not need to establish these gateways. Refer to "VoIP interoperability: Gatekeeper configuration" on page 417.

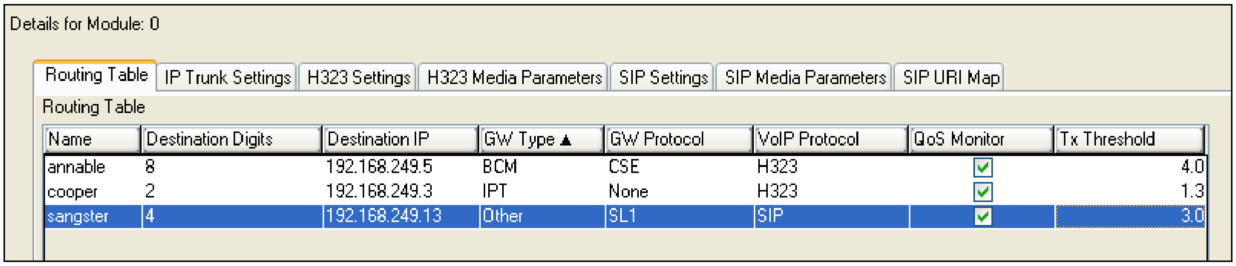
The following path indicates where to access the remote gateway in Element Manager:

Element Manager: **Configuration > Resources > Telephony Resources > Routing Table**

To configure a remote gateway

1. On the Modules panel, in the Module type column, select the IP Trunks line.
2. In the bottom panel, select the **Routing Table** tab.
3. Click **Add**.
4. The **Add Remote Gateway** dialog box appears. See Figure 129.
5. Enter a Name and Destination Digits for the remote gateway.
6. Click **OK**.
7. Enter the appropriate information for the remote system.

Figure 129 Remote gateway record



Routing Table

Name	Destination Digits	Destination IP	GW Type ▲	GW Protocol	VoIP Protocol	QoS Monitor	Tx Threshold
annable	8	192.168.249.5	BCM	CSE	H323	☑	4.0
cooper	2	192.168.249.3	IPT	None	H323	☑	1.3
sangster	4	192.168.249.13	Other	SL1	SIP	☑	3.0

Table 5 Remote gateway record

Field	Value	Description
Name	<alphanumeric>	Enter an identifying tag for the remote system.
Destination Digits	<numeric>	Enter digits that identify the remote system as call destination
Destination IP	<IP address>	Indicate the IP address of the device you want to connect with. This code will be part of your destination code programming.
GW Type	BCM BCM35 IPT Other	Choose the variable that identifies the type of system or application being connected to.
GW Protocol	None SL1 SCE	Choose the protocol that supplies the required call features. None (default) supplies no feature. This setting is dictated by the type of remote system.
VoIP Protocol	SIP H323	Choose signaling to endpoint protocol.
QoS Monitor	<check box>	Enable this feature if you are using fallback to PSTN lines and the network supports QoS monitoring.
Tx Threshold	0.0 (bad) 5.0 (excellent)	Indicate the level of transmission at which the signal must be maintained. If the signal falls below this level the call falls back to PSTN.

Configuring VoIP lines

VoIP lines require a keycode to activate. You also need to set gateway parameters and system IP parameters to enable the trunks.

You must also set up target lines when you use these trunks.

The following path indicates where to set up target lines in Element Manager:

- Element Manager: Configuration > Telephony > Lines > Target Lines

Prerequisites

Complete the following prerequisites checklist before configuring the modules.

The gateway and IP network is set up correctly. Refer to the following procedures: • “Configuring VoIP trunk media parameters” on page 410 • “Setting up the local gateway” on page 410 • “Setting up remote gateways” on page 412 • “VoIP interoperability: Gatekeeper configuration” on page 417	
Obtain all relevant central office/service provider information for the type of trunk.	

Configuring VoIP line features

The following procedure describes the fields that need to be confirmed or set for these lines. For detailed field descriptions, refer to “Configuring lines” on page 147.

1 Confirm or change the settings on the Line/Trunk main panel:

- Line: Unique number
- Trunk type: VoIP
- Name: identify the line or line function
- Control Set: identify a DN if you are using this line with scheduling.
- Line Type: define how the line will be used. If you are using routing, ensure it is put into Pool (A to F)
- Prime Set: If you want the line to be answered at another telephone if the line is not answered at the target telephone, otherwise, choose None.
- Pub. Received #: Not applicable
- Priv. Received #: Not applicable
- Distinct Ring: If you want this line to have a special ring, indicate a pattern (2, 3, 4, or None).

2 On the bottom panel, under the restrictions tab:

- Use remote package: If this line is used for remote call-ins or is part of a private network, ensure you specify a valid remote package.

3 Configure the trunk/line data:

In the top panel ensure a loop trunk is selected. In the bottom panel, select the preferences tab.

- Aux. ringer: If your system is equipped with an external ringer, you can enable this setting to allow this line to ring at the external ringer.

4 Set the restriction and remote restrictions scheduling (Restrictions tab):

- Line Restrictions: Enter a valid restriction filter for the Normal schedule, and any other schedules that you want this line to be part of. (outgoing calls)
- Remote Restrictions: Enter a valid remote access package for the Normal schedule, and any other schedules that you want this line to be part of. (incoming calls from remote users or private networks)

5 Suggested next steps:

- "Configuring lines: Target lines" on page 177"Configuring lines: Target lines and DASS2" in the *BCM 4.0 Device Configuration Guide* (N0060600)
- Also refer to "Line Access - Line Pool Access tab" in the *BCM 4.0 Device Configuration Guide* (N0060600)
- "Dialing plan: Routing and destination codes" on page 289
- "Dialing plan: Private network settings" on page 317

Chapter 46
VoIP interoperability: Gatekeeper configuration

The following describes the use of a gatekeeper for your H.323 VoIP trunks.

Refer to the gatekeeper software documentation for information about changing IP addresses.

Gatekeeper notes:

- The BCM has been tested by Nortel to be compliant with CS 1000 gatekeeper applications.
- A gatekeeper may help to simplify IP configuration or the BCM dialing plan; however, it does not simplify the network dialing plan.

Using CS 1000 as a gatekeeper

Both the BCM and the CS 1000 must be set to the parameters described in the following information for the gatekeeper to work effectively. Refer to *IP Peer Networking: Installation and Configuration* (553-3001-213) for detailed information on configuring a CS 1000 gateway.

For CS 1000, the Network Routing Service (NRS) can be configured and maintained through a web interface called NRS Manager. NRS Manager replaces the CS 1000 GK admin tool.

Review the following information before attempting to use the CS 1000 as a gatekeeper:

- Before a Gateway Endpoint registers with the CS 1000 gatekeeper it must first be added to the gatekeeper configuration.
- Before a registered Gateway Endpoint makes calls, it must have its routing entry information assigned within the gatekeeper configuration.
- Before any of these configuration changes become part of the gatekeeper active configuration, they must be committed to the active database.

BCM requirements

Set the BCM Local Gateway IP interface to the following using BCM Element Manager (go to **Configuration > Resources > Telephony Resources > {Select IP Trunk} > H323 Settings tab**):

- Set **Call Signaling** to GatekeeperRouted or GatekeeperResolved.
- Set **Primary Gatekeeper IP** to the IP address of the NRS.
- Set **Alias Names** to the Alias name that was used when the H.323 Endpoint for the BCM was created on the NRS.

In order to make a BCM 3.01 (or later)-to-CS 1000 call, ensure that the BCM routes and dialing plan (used to reach the CS 1000 systems) match the numbering plan entry assigned to the CS 1000 systems through NRS Manager.

Similarly, to make a CS 1000 system-to-BCM 3.01 (or later) call, ensure that the numbering plan entry assigned to the BCM (through NRS Manager) matches the dialing plan information configured on the CS 1000 systems.

CS 1000 configuration

You must use NRS Manager to configure the CS 1000.

The NRS server must be enabled and properly configured before any NRS data can be provisioned using NRS Manager. Refer to *IP Peer Networking: Installation and Configuration* (553-3001-213) for detailed information on configuring a CS 1000 gateway.

Chapter 47
T.38 fax

If you are using the T.38 fax protocol, it is assumed that you have already configured IP trunks and gateways, and that they are functional. For more information on configuring VoIP trunks see "Configuring lines" on page 147.

T.38 fax is a Fax over IP (FoIP) gateway protocol that allows standard (T.30 or Group3) fax machines to make calls over IP-based networks. The T.38 fax protocol functions transparently with standard fax machines because it emulates a normal T.30 fax connection. Each endpoint of the IP trunk becomes a T.38 gateway. To use FoIP, you must have two or four MS-PEC III cards installed in your MSC card. Both endpoints must support the T.38 fax protocol and have this feature enabled.

Enabling T.38 fax

Complete these procedures to enable the T.38 fax protocol.

To verify codecs in Element Manager

1. Click **Configuration > Telephony Resources**.
2. In the Telephony Resources panel, select the row for IP Trunks on Bus 0.
   The details panel appears.
3. Click the **H323 Media Parameters** or the **SIP Media Parameters** tab.
4. Verify that the preferred codec appears in the **Selected List** field.

5 Verify that the codecs are set at the default before performing T.38 sessions.

To enable a T.38 fax

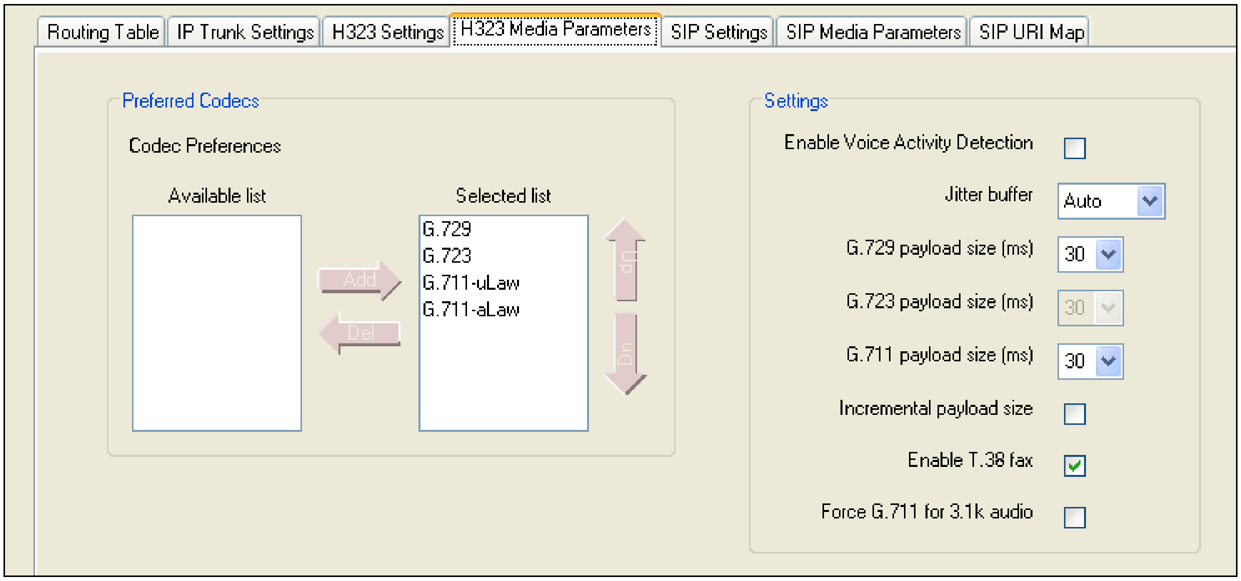
1 Click **Configuration > Telephony Resources**.

2 In the Telephony Resources panel, select the row for IP Trunks on Bus 0. The details panel appears.

3 Click the **H323 Media Parameters** tab or the **SIP Media Parameters** tab.

4 Select the **Enable T.38 fax** check box.

Figure 130 H323 Media Parameters tab



Selected list: G.729, G.723, G.711-uLaw, G.711-aLaw

Enable Voice Activity Detection □

Jitter buffer: Auto

G.729 payload size (ms): 30

G.723 payload size (ms): 30

G.711 payload size (ms): 30

Incremental payload size □

Enable T.38 fax ☑

Force G.711 for 3.1k audio □

Lines

To enable T.38 fax protocol you must configure the following:

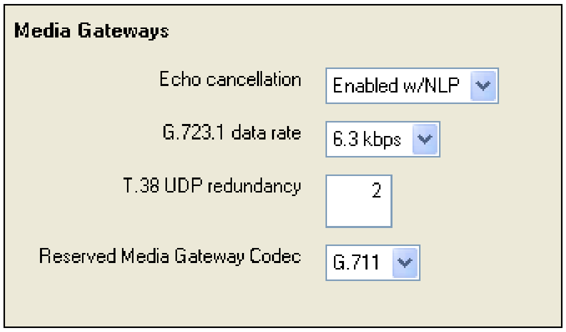
- Voice over IP (VoIP) lines (see "Configuring lines" on page 147)
- target lines (see "Configuring lines: Target lines" on page 177)
- call routing (see "Dialing plan: Routing configurations" on page 277)
- destination codes (see "Destination codes" on page 292)

Media gateways

T.38 UDP redundancy refers to the number of times IP packets (not fax pages) are sent, because TCP/UDP does not support packet validation (unlike TCP/IP).

To configure media gateways, click **Configuration > Resources > Media Gateways**.

Figure 131 Media Gateways panel



Media Gateways

Echo cancellation: Enabled w/NLP

G.723.1 data rate: 6.3 kbps

T.38 UDP redundancy: 2

Reserved Media Gateway Codec: G.711

Note: For more details and instructions on how to configure media gateways, see "Media Gateways" on page 439.

T.38 Fax restrictions

Note: Fax tones that broadcast through a telephone speaker can disrupt calls on other telephones using VoIP trunks near the fax machine. Follow these suggestions to reduce the chance of your VoIP calls being dropped because of fax tone interference:

- Locate the fax machine away from other telephones.
- Turn the speaker volume on the fax machine to the lowest level, or off.

Note: Fax tones can be recorded in a voice mail box. In the rare event that fax tones are captured in a voice mail message, opening that message from a telephone using a VoIP trunk can cause the connection to fail.

Voice mail, IVR, and T.38 FoIP share a maximum of eight fax ports. Voice mail supports only two fax ports.

If you allow fax messaging for the local VoIP gateway, you must be aware of the guidelines in "Operational notes and restrictions" on page 421 when you send and receive fax messages over VoIP trunks. For more information, see "VoIP trunk gateways" on page 395.

Operational notes and restrictions

Some fax machines cannot send faxes successfully over VoIP (T.38) trunks to the following destinations:

- CallPilot mailboxes
- CallPilot mailboxes accessed through auto-attendant
- Fax Transfer (calls transferred to a system fax device through the auto-attendant)

Use the following tips to avoid this problem:

- Avoid using manual dial on the originating fax machine. In some fax machines, dialing manually results in a much shorter call time-out.
- If you must dial manually, wait until the call is answered before you start the fax session.
- For Mailbox Call Answering only, if you must dial manually, enter the digit 8 as soon as you hear the mailbox greeting. This ensures that CallPilot initiates the fax session before the fax machine timer starts.

Note: Enter the digit 8 for Norstar Voice Mail User Interface (NVMUI) only. To enable fax call answering when using CallPilot User Interface (CPUI), enter 707.

- Increase the call duration by adding a timed pause to the end of the dialing string. This addition allows the call to ring at the destination before the fax machine call-duration timer starts. Refer to your fax machine documentation for more information on how to insert pauses into dial strings.
- Because the problem is related to the delay in initiating the fax session, reduce the number of rings for fax mailboxes Call Forward No Answer (CFNA).

Chapter 48
Setting up VoIP trunks for fallback

The following path indicates where to access setting VoIP trunks for fallback in the Element Manager:

Element Manager: **Configuration > Resources > Telephony Resources > Local Gateway tab** (bottom panel)

Task: Configure VoIP trunks to allow fallback to PSTN lines

- “Configuring routes for fallback” on page 423
- “Example: A private network configured for fallback” on page 428

Configuring routes for fallback

Configuring routes allows you to set up access to the VoIP and the PSTN line pools. These routes can be assigned to destination codes. The destination codes then are configured into schedules, where the PSTN line is assigned to the Normal schedule and the VoIP route is assigned to a schedule that can be activated from a control set.

For details about route and schedule configuration, refer to the information under the headings below:

- “Adding routes for fallback” on page 424
- “Assigning the line pools to routes” on page 424
- “Adding the destination code for the fallback route” on page 425
- “Configuring the schedules for the destination codes” on page 426
- “Setting up the VoIP schedule to overflow” on page 427

Pre-configuration requirements

- If you have not already done so, remember to define a route for the local PSTN for your own system so users can still dial local PSTN numbers.
- Ensure the PSTN and VoIP line pools have been configured before you continue with this section. For information about creating a VoIP line pool, see “Configuring VoIP trunk gateways” on page 409. Configure PSTN lines under **Configuration > Telephony > Lines > Active Physical Lines**.

➡ **Note:** If you already have routes for your PSTN or VoIP line pools configured, you do not need to configure new routes, unless you cannot match the dialed digits.

Adding routes for fallback

Enter the routes you want to use for normal and fallback traffic.

Add routes under **Configuration > Telephony > Dialing Plan > Routing**.

To add the PSTN route (to other system)

1. Type a number between 001 and 999.
   This route defines the PSTN route to the other system. Only numbers not otherwise assigned will be allowed by the system.
2. Click **OK**.

To add the PSTN route to the local PSTN lines

1. In the **Route** field, type a number between 001 and 999.
   This route defines the PSTN route to your local PSTN.
2. Click **Save**.

To add the VoIP route

1. In the **Route** field, type a number between 001 and 999.
   This route defines the VoIP route.
2. Click **Save**.

Assigning the line pools to routes

Assign the line pools to the routes you created in the "Line pools (and access codes)" on page 297.

To assign PSTN line pool (to other system)

1. Click the route you created between the PSTN line and the other system.
2. In the **Use Pool** box, type the letter of the line pool for the PSTN lines to the other system.
3. In the **External Number** field:
   If this is a public PSTN line, enter the dial numbers that access the other system through the PSTN. For example: 1<area code><local code>.
4. In the **DN Type** box, choose **Public**.

To assign PSTN line pool to local PSTN lines

1. Click the route you created for your local PSTN line.
2. In the **Use Pool** box, type the letter of the line pool for the PSTN line.
3. In the **External Number** field: leave this field blank.
4. In the **DN Type** box, choose **Public**.

To assign a VoIP line pool

1 Click the route you created for the VoIP lines.

2 In the **Use Pool** field, type the letter of the line pool for the VoIP lines.

3 Leave the **External Number** field blank unless the destination digit you are using for the remote gateway is different than the number you want to use for the destination code.

4 In the **DN Type** box, choose **Private**.

Go to the next section: "Adding the destination code for the fallback route" on page 425.

Adding the destination code for the fallback route

Create a destination code that includes the VoIP and PSTN routes that you created in "Adding routes for fallback" on page 424 to respond to the same access number (destination code). When this code is dialed, the BCM will select the VoIP line, if possible. If the line is not available, the call will fall back to the PSTN line.

As well, you need to create, or ensure, that your destination code 9 includes a Normal and VoIP schedule that includes the route you created to the local PSTN.

➡ **Note:** If you already have a line pool access code defined as 9, you will need to delete this record before you create the destination code.

To create destination codes for your fallback route

1 Select **Configuration > Telephony > Dialing Plan > Routing**, and then select the **Routes** tab.

2 Click **Add**.
The Add Route dialog box appears.

3 Enter one or more digits for this destination code.

4 Click **OK** to close the dialog box.

Example:

Destination code digit: If it is available, you might want to use the same number that you used for the destination code of the gateway.
If you have multiple gateways, you could use a unique first number followed by the destination digits, to provide some consistency, such as 82, 83, 84, 85 to reach gateways with destinations digits of 2, 3, 4 and 5.
The number you choose will also depend on the type of dialing plan the network is using. Networks with CDP dialing plans have unique system codes. However, with networks using UDP, this is not always the case, therefore, you need to be careful with the routing to ensure that the codes you choose are unique to the route. This will also affect the number of digits that have to be added or absorbed. It is helpful to use the Programming Records to plan network routing so you can determine if there will be any conflicts with the destination codes you want to use.

Configuring the schedules for the destination codes

Under the destination code heading you created in the previous section, click the **Schedules** key, then choose the appropriate schedules:

To configure the VoIP schedule for all fallback destination codes

1. Change **First Route** to the route you configured for your VoIP line.
2. Set the **Absorbed length** to absorb the amount of the destination code that is not part of the dialout for the trunk.

Normal schedule for all fallback destination codes:

1. Change **Use Route** to the route you configured for your PSTN fallback line (the line to the other system).
2. Set the **Absorbed length** to absorb the amount of the destination code that is not part of the DN for the other system.

Examples:

Absorbed length, VoIP schedule: If the remote gateway destination digit is 2, which is part of the remote system DN structure (CDP network), and you specified a destination code of 82, set this field to 1, so that the 2 is still part of the dialout.
If the destination code is different from the remote gateway destination digits, and you entered an External # into the route record (the destination digit for the remote system), set the absorbed length to the number of digits in the destination code. The system will dial out the External # you entered in front of the rest of the number that the user dialed. This would occur if the network is set up with a UDP dialing plan.

Note: Do not add alternative routes (second or third). Since fallback is active, the system immediately falls back to the Normal schedule if the first route is not available.

Absorbed length, Normal schedule: If this is a private network PSTN line, and the network uses a CDP dialing plan, and the remote system identifier is 2, which is part of the remote system DN structure, and you specified destination digit of 2 for the remote gateway, then configured a destination code of 82, set this field to 1, so that the 2 is still part of the dialout.
If the destination code is different from the private access code/destination digits for the remote system (UDP dialing plan) or this is a public PSTN, enter private access code or the public access number to the remote system into the External # field on the route record. In this case, set the absorbed length to the number of digits in the destination code. The system will dial out the External # you entered in front of the rest of the number that the user dialed.

Setting up the VoIP schedule to overflow

Once you have configured the routing and destination codes, ensure that the Routing Service schedule allows fallback (Overflow) and allows you to activate the service from a control set. You will note that the Routing Service does not have a Normal schedule. This is because the Normal schedule is the schedule that runs when no routing services are active.

To set up the VoIP schedule for routing services

1. Double-click Sched 4 and rename it **VoIP** (**Configuration > Telephony > Scheduled Service > Schedule Column**).
2. Click **VoIP**.
   The VoIP schedule panel appears in the right frame.
3. Change the **Routing Svc** to **Manual**.
4. Select the **Overflow** check box.
5. Next steps:

   The following topics describe some further actions you may need to take to ensure that fallback is working:

   — “Activating the VoIP schedule for fallback” on page 427
   — “Deactivating the VoIP schedule” on page 428

Activating the VoIP schedule for fallback

Before activating the VoIP schedule, calls using the destination code are routed over the PSTN. This is because the system is set to use the Normal schedule, which routes the call over the PSTN. Once the VoIP schedule is activated, calls made with the VoIP destination code are routed over the VoIP trunk.

The VoIP line must be activated (**FEATURE 873**) from the control set for the VoIP trunk, which is specified when the trunk is created (**Configuration > Telephony > Lines > Active VoIP Lines**).

To activate the VoIP line from the control set

1. Dial **FEATURE 873** from the control set for the VoIP trunk.
   The phone prompts you for a password.
2. Type the password (default - admin: 23646).
3. Press **OK**.
   The first schedule appears.
4. Scroll down the list until VoIP is selected.
5. Press **OK**.
   The VoIP schedule stays active, even after a system reboot, and can only be manually deactivated.

Deactivating the VoIP schedule

To deactivate a schedule

1. Dial **FEATURE #873**. The phone prompts you for a password.
2. Type the password.
3. Press **OK**. The system returns to the Normal schedule.

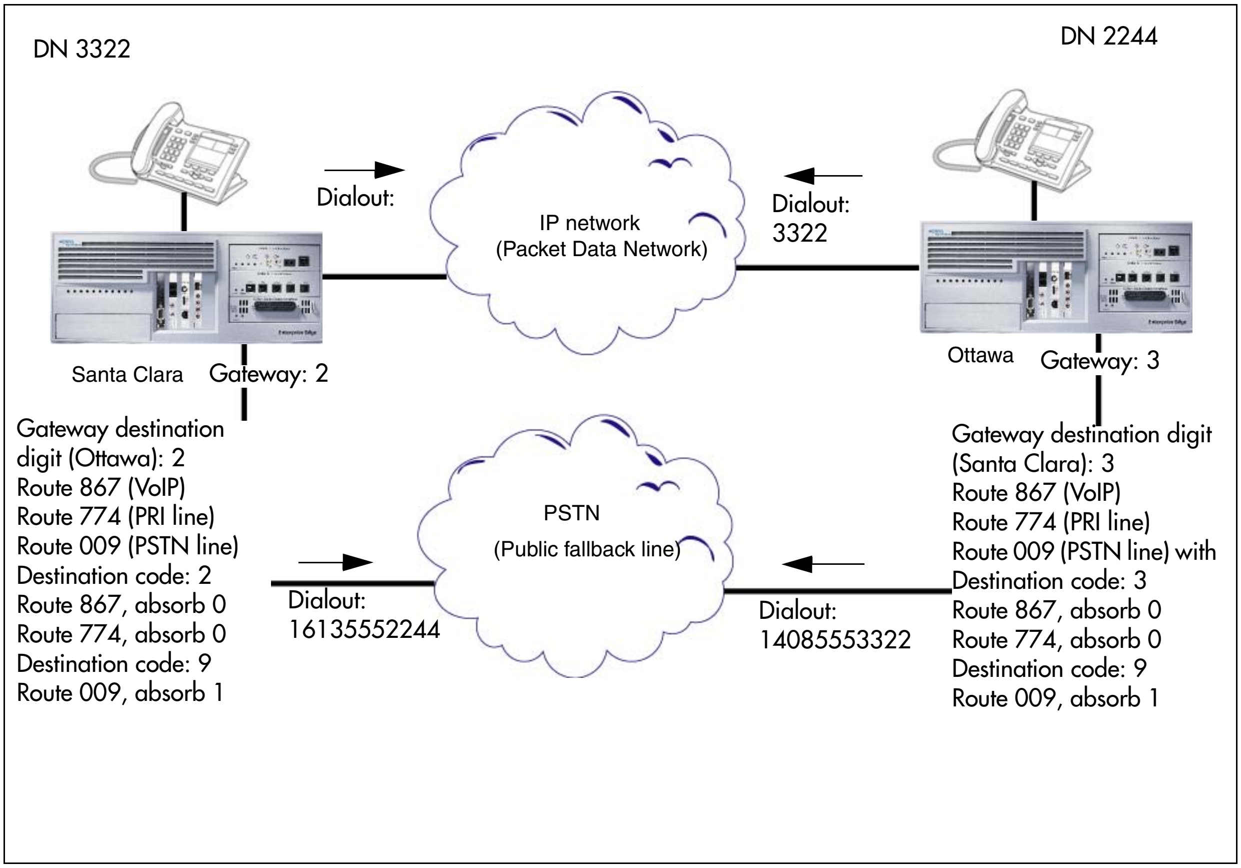
Example: A private network configured for fallback

The following example explains through a sample BCM configuration, including:

- "Activating the VoIP schedule for fallback" on page 427
- "Deactivating the VoIP schedule" on page 428

In this scenario, shown in Figure 132, two BCMs in different cities are connected through a WAN. One BCM resides in Ottawa, the other resides in Santa Clara. Both VoIP trunks and an PRI SL-1 line connect the system in a private network.

Figure 132 Example PSTN fallback



BCM Santa Clara	BCM Ottawa
• IP address: 47.62.84.1	• IP address: 47.62.54.1
• DNs 3000-3999	• DNs 2000-2999
• From this system, dial 9 to get onto PSTN	• From this system, dial 9 to get onto PSTN
• Dialing plan: CDP	• Dialing plan: CDP, destination code is part of DN

Routing	Routing
• Target DN 2244 (first digit is unique to system)	• Target DN 3322 (first digit is unique to system)
• Remote gateway destination digit: 2	• Remote gateway destination digit: 3
• Destination code: 2	• Destination code: 3
• VoIP/private network dialout: no external #, user dials 2244 (no absorbed digits)	• VoIP/private network dialout: no external #, user dials 3322 (no absorbed digits)

The systems already communicate through a PRI line, which will be configured to be used for fallback. Both systems already have all keycodes installed for eight VoIP lines, and resources properly allocated for VoIP trunking. For information about keycodes, see the *Keycode Installation Guide* (N0060625).

Each BCM has 10 telephones that will be using VoIP lines. In this setup only eight calls can be sent or received over the VoIP trunks at one time. If all 10 telephones attempt to call at the same time, two of the calls will be rerouted to the PSTN or other alternate routes if multiple routing is set up in the destination code schedule.

System programming for networking and fallback routes

Table 98 provides the settings that are required for both systems to create a fallback network.

Table 98 Fallback configuration to create fallback between two systems (Sheet 1 of 2)

Task	Settings for Santa Clara	Settings for Ottawa	Location in Element Manager
Set up a Control set for each VoIP line	3321	2221	**Configuration > Telephony > All DNs**
Set first preferred Codec	G.729		**Configuration > Resources > Telephony Resources > IP Trunks, H.323 Trunks, Media Parameters** tab.
Set voice activity detection	Selected		
Set Jitter Buffer	Medium		
Put 8 VoIP lines into the same line pool	Pool A through Pool O		
Give all system telephones access to the VoIP line pool	BlocO		**Configuration > Telephony > Dialing Plan > Line Pools**

Table 98 Fallback configuration to create fallback between two systems (Sheet 2 of 2)

Task	Settings for Santa Clara	Settings for Ottawa	Location in Element Manager
Confirm or assign target lines to all DNs or Hunt Groups.	<targetline #>		**Configuration > Telephony > Lines > Target Lines**
Configure the target lines that you assigned.	Control set: 3321	Control set: 2221	**Configuration > Telephony > Lines > Target Lines > Line XXX**
	Trunk/Line data: Line Type: Private If busy: To prime		
	Prime set: DN 3321 Received number: 3322	Prime set: DN 2221 Received number: 2244	
Create remote gateway record for remote BCM	Destination IP: 47.62.54.1	Destination IP: 47.62.84.1	**Configuration > Resources > Telephony Resources > IP Trunks**, Remote Gateway
	QoS Monitor: Enabled Transmit Threshold: 3.5 (moderate quality) Gateway Type: BCM3.6 Gateway protocol: None		
	Destination Digits (Ottawa): 2	Destination Digits (Santa Clara): 3	**Destination digits note:** In this case, the systems use a Coordinated Dialing Plan (CDP) network, and the destination digit is included in the DN.
Set up Scheduling to allow you to manually start and stop schedules.	Service setting: Manual Overflow: Selected		**Configuration > Telephony > Scheduled Services**, VoIP (Schedule 4).
Confirm or set up a route using the line pool to access the local PSTN.	Route: 009		**Configuration > Telephony > Dialing Plan > Routing**.
	External #	External #	
	Line Pool: <publiclinepool> DN type: Public		
Set up a route that contains the PRI fallback lines.	Route: 774 Dialout: N/A PSTN Line Pool: BlocA DN type: Private		**Configuration > Telephony > Dialing Plan > Routing.**
Set up a route that contains the VoIP line pool.	Route: 867 Dialout: N/A VoIP Line: BlocF DN type: Private		**Configuration > Telephony > Dialing Plan > Routing.**
Create a destination code that matches the Destination Digit(s).	Destination code: 2	Destination code: 3	
Define the Normal and VoIP shedules.	Normal: Route 774, Absorb 0 digits VoIP: Route 867, Absorb 0 digits		**Configuration > Telephony > Scheduled Services**
Confirm or create a destination code for the PSTN. Define Normal and VoIP schedules.	Destination code: 9 Normal: Route 009, absorb All digits VoIP: Route 009, absorb All digits		
Activate the VoIP schedule from the control set.	3321	2221	**FEATURE 873**

Making calls through a private VoIP network gateway

From a telephone on BCM Ottawa, a caller dialing to a telephone on BCM Santa Clara must dial the destination code, which includes the destination digits for the BCM Santa Clara remote gateway, and the DN of the telephone. For example, dialing 3322 would connect as follows:

- 3 is the destination code. If a suitable level of QoS is available, the call is routed through the VoIP trunks and through the remote gateway with a destination digit of 3. The call is sent across the PDN using the IP address of the Santa Clara BCM.
- 3322 is linked to the target line associated with DN 3322.
- The call arrives at the phone with the DN 3322.

If a user in Santa Clara wanted to make a local call in Ottawa, they would dial 29, followed by the local Ottawa number. The digit 2 accesses the remote gateway for the VoIP line. The digit 9 accesses an Ottawa outside line.

Chapter 49

Port ranges overview

The Port Ranges panel provides a list of which Ports are currently being used for RTP/UDP, UDP, and Signaling. In the case of RTP over UDP and UDP, it allows changes to the ports being used.

For information on configuring port ranges, see "Port Ranges Panel" on page 435.

⚠ **Warning:** Port configuration should not be changed unless absolutely necessary, such as in instances where port configurations are causing conflicts, or if a firewall is restricting communications over certain ports.

RTP over UDP

RTP over UDP is used by IP sets to connect to media gateways, and by IP trunks to connect to remote devices or PDM devices. All of these services require RTP over UDP. Each media gateway uses two ports. By default, RTP over UDP is set to use the port range 28000 - 28255. Nortel recommends that you keep 256 ports configured for RTP over UDP. The BCM requires a minimum of 206 ports to support necessary services. This includes 32 IP sets, 11 voice mail and contact center voiceports, and 60 trunks. Each of these devices requires two RTP over UDP ports.

You can configure up to ten separate ranges of ports.

UDP

UDP is used for T.38 Fax over UDP. By default, it uses the Range 20000 to 20255. You can configure up to ten separate ranges of ports. While the system can function with 12 ports, it is recommended that 256 ports are reserved.

Signaling Ports

Signaling ports are used by the system and cannot be modified. They are provided to show where conflicts with UDP or RTP occur.

Chapter 50
Port Ranges Panel

The Port Ranges panel allows you to reserve ports for use by UDP (User Datagram Protocol). The Port Ranges panel consists of three tables: RDP over UDP, UDP, and Signaling.

Panel tabs	Tasks	Features
"RTP over UDP Port Ranges" on page 435	"Port Ranges panel" on page 436	
"UDP Port Ranges" on page 437	"Deleting RTP over UDP Port Ranges" on page 436	
"Signaling Port Ranges" on page 438	"Modifying RTP over UDP Port Ranges" on page 437	
	"UDP (User Datagram Protocol) ports are necessary for certain types of network communications. The UDP table has two settings, as shown in Table 100." on page 437	
	"Deleting UDP Port Ranges" on page 437	
	"To modify an entry on the UDP table" on page 438	

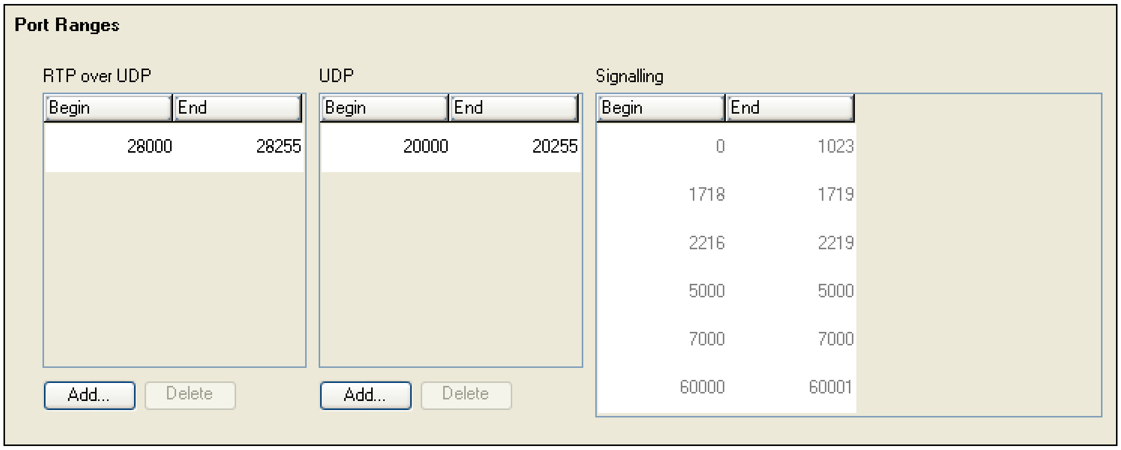
⚠ **Warning:** Do not change the ports unless necessary. If you do change the ports, make sure you review the minimum requirements for each protocol. As well, make sure that you configure your firewall to reflect any changes you make to the ports.

RTP over UDP Port Ranges

RTP (Real-time Transfer Protocol) over UDP ports are necessary for IP trunk traffic, such as for the transmission of audio and video signals across the Internet. These values should only be changed if you are interoperating with an unsupported product. The RTP over UDP table has two settings.

Figure 133 illustrates the Port Ranges panel.

Figure 133 Port Ranges panel



Port Ranges

RTP over UDP

Begin	End
28000	28255

UDP

Begin	End
20000	20255

Signalling

Begin	End
0	1023
1718	1719
2216	2219
5000	5000
7000	7000
60000	60001

Table 99 RTP over UDP

Attribute	Value	Description
Begin	<numeric string>	The first port in the port range.
End	<numeric string>	The last port in the port range.

Note: You can add up to ten port ranges.

To add new port ranges in the RTP over UDP table

1. On the **RTP over UDP** table, click **Add**.
   The **Add RTP Port Range** dialog box appears.
2. In the **Begin** field, enter the first port in the range.
3. In the **End** field, enter the last port in the range.
4. Click **OK**.
   The new RTP port range appears in the table.

Deleting RTP over UDP Port Ranges

You cannot delete all port ranges from the table. You must keep at least one port range at all times.

To delete port ranges from the RTP over UDP table

1. On the **RTP over UDP** table, select the range to delete by clicking the appropriate row in either column.
2. Click **Delete**.
   A confirmation dialog box appears.
3. Click **Yes**.

Modifying RTP over UDP Port Ranges

To modify an entry on the RTP over UDP table

1. On the **RTP over UDP** table, select the entry to modify.
2. Type in the new value.

UDP Port Ranges

UDP (User Datagram Protocol) ports are necessary for certain types of network communications. The UDP table has two settings, as shown in Table 100.

Table 100 UDP

Attribute	Value	Description
Begin	<numeric string>	The first port in the port range.
End	<numeric string>	The last port in the port range.

➡ **Note:** You can add up to ten port ranges.

To add new port ranges in the UDP table

1. On the **UDP** table, click **Add**.
   The **Add UDP Port Range** dialog box appears.
2. In the **Begin** field, enter the first port in the range.
3. In the **End** field, enter the last port in the range.
4. Click **OK**.
   The new RTP port range appears in the table.

Deleting UDP Port Ranges

➡ **Note:** You cannot delete all port ranges from the table. You must keep at least one port range at all times.

To delete port ranges from the UDP table

1. On the **UDP** table, select the range to delete.
2. Click **Delete**.
   A confirmation dialog box appears.
3. Click **Yes**.

To modify an entry on the UDP table

1 On the **UDP** table, select the entry to modify.

2 Type in the new value.

Signaling Port Ranges

Table 101 displays port ranges used for signaling. These port ranges cannot be modified. The Signaling Port Ranges table consists of two fields:

Table 101 Signaling

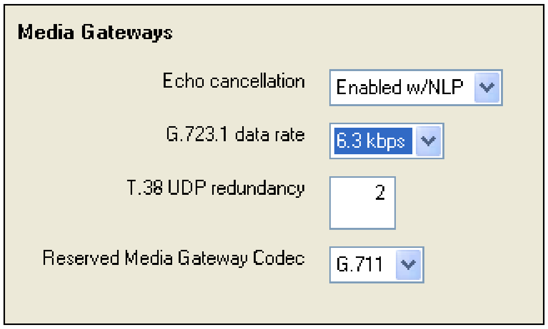
Attribute	Value	Description
Begin	<read-only numeric string>	The first port in the port range.
End	<read-only numeric string>	The last port in the port range.

Chapter 51
Media Gateways

Certain types of IP communications pass through Media Gateways on the BCM. You can control the performance of these communications by adjusting the parameters for echo-cancellation and UDP Redundancy.

The Media Gateways panel allows you to set basic parameters that control IP telephony. Figure 134 shows the fields on the Media Gateways panel.

Figure 134 Media Gateways



Media Gateways

Echo cancellation: Enabled w/NLP

G.723.1 data rate: 6.3 kbps

T.38 UDP redundancy: 2

Reserved Media Gateway Codec: G.711

Table 102 General Settings

Attribute	Value	Description
Echo cancellation	<drop-down menu> Enabled w/NLP Enabled Disabled	Enable or disable echo cancellation for your system. Default: Enabled w/NLP (check with your internet system administrator before changing this) **Echo Cancellation** selects what type of echo cancellation is used on calls that go through a Media Gateway. NLP refers to Non-Linear Processing.
G.723.1 data rate	6.3 kbps 5.3 kbps	Specify the data rate at which to process calls. This setting is not negotiated; it is manually administered and the far end must support the rate. The rate selection is based on quantity of calls, or quality of voice. • 5.3 kbps is a smaller bandwidth therefore more calls can be supported but the voice quality is lower. • 6.3 kbps is a larger bandwidth so fewer calls are supported but the voice quality is better. Default: 6.3 kbps
T.38 UDP redundancy	<numeric character string>	If T.38 fax is enabled on the system, this setting defines how many times the message is resent during a transmission, to avoid errors caused by lost T.38 messages.
Reserved Media Gateway Codec	<protocol>	Select a protocol to compress and transmit packets. G.711 G.729 G.723

Chapter 52

Call security and remote access

System restrictions are required to ensure that your system is used appropriately and not vulnerable to unauthorized use.

Call security includes:

- restriction filters, which limit outbound call access
- remote access packages, which limit system call feature access for users calling in over the Private or Public network
- Class of Service codes, which require remote system users to enter a password before they can access the system. CoS passwords also can have restriction filters applied.

These topics are discussed under the following sections:

Call security works in conjunction with your dialing plan. Refer to "Dialing plans" on page 247.

Defining restriction filters

Restriction filters allow you to restrict the numbers that can be dialed on any external line within BCM. Up to 100 restriction filters can be created for the system.

To restrict dialing within the system, you can apply restriction filters to:

- outgoing external lines (as line restrictions)
- telephones (as set restrictions)
- external lines on specific telephones (as line/set restrictions)

Restriction filters can also be specified in Restrictions service for times when the system is operating according to a schedule. Dialed digits must pass both the line restrictions and the set restrictions. The line per set (line/set) restriction overrides the line restriction and set restriction.

Notes about restriction filters

A restriction filter is a group of restrictions and overrides that specify the external numbers or feature codes that cannot be dialed from a telephone or on a line. The restriction filters setting allows you to assign restrictions in one step as a single package of dialing sequences that are not permitted.

In addition to restricting telephone numbers, you can prevent people from entering dialing sequences used by the central office (the public network) to deliver special services and features. Some of these features provide the caller with dial tone after they have entered the special code (which often uses # or *), therefore, users have an opportunity to bypass restrictions. To prevent this from happening, you can create filters that block these special codes.

You create a filter by defining the dialing sequences that are denied. There are also variations of each sequence that you want users to be able to dial, these are called overrides. Overrides are defined within each restriction package for each filter.

Once you create the filters, you can assign the restrictions to a telephone, to a line, to a particular line on a telephone, and to remote callers.

➡ **Note:** Filter 00 cannot be changed. Filter 01 has a set of defaults. Filters 02 to 99 can be set to suit your special requirements. See "Default filters (North America)" on page 442.

- Each programmable filter can have up to 48 restrictions.
- There is no limit on the number of overrides that can be allocated to a restriction. However, there is a maximum total of 400 restrictions and overrides allocated to the 100 programmable filters.
- The maximum length of a restriction is 15 digits.
- The maximum length of an override is 16 digits.
- Entering the letter *A* in a dialing sequence indicates a wild card, and represents any digit from 0 to 9.
- You can use * and # in a sequence of numbers in either a restriction or an override. These characters are often used as part of feature codes for other systems or for features provided by the central office (the public network).
- When restricting the dialing of a central office feature code, do not forget to create separate restrictions for the codes used for DTMF and pulse lines (for example, *67 and 1167).
- Do not string together a central office feature code and a dialing sequence that you want to restrict. Create a separate restriction for each.
- You can copy restrictions and overrides from one filter to another. You can use a restriction or override in any number of filters. Each time you use a restriction or override, it counts as one entry. For example, if restriction 411 exists in filters 01, 02 and 03, it uses up three entries of the 400 entries available.
- Removing a restriction from a filter has no effect on the contents of other filters, even if the restriction was copied to them.
- You cannot delete a filter. Removing the restrictions programmed on a filter makes it an unrestricted filter but the filter itself is not removed.

Default filters (North America)

Filter 00 permits unrestricted dialing and cannot be changed.

Filter 01 is pre-programmed with 10 restrictions and some associated overrides. In Filter 01, Restriction 02 and Override 001 allow long distance toll free calls.

The dialing string 911, which is the number for emergency assistance in North America, is included as both a restriction and an override in Filter 01. This arrangement prevents anyone from blocking calls for emergency assistance on lines or sets using the default filter.

Table 103 Default restriction filters

Filter	Restrictions (denied)	Overrides
00	Unrestricted dialing	
01	01: 0	
	02: 1	001: 1800 002: 1877 003: 1888
	03: 911	001: 911
	04: 411	
	05: 976	
	06: 1976	
	07: 1AAA976	
01	08: 1900	
	09: 1AAA900	
	10: 5551212	
02 - 99	No restrictions or exceptions programmed	

Note: Default filters are loaded when the system is initialized. A cold start restores the default filters.

Filters 02, 03, and 04, although not preset with restrictions and overrides, are the default filters in these programming headings:

Table 104 Default filters for program headings

Filter	Heading	Sub-heading
02	System DNs	Set restrictions
03	Lines	Line restriction
04	Lines	Remote restriction

Default filters (other)

Three profiles have global overrides which do not appear in Element Manager restriction programming and cannot be changed.

Australia: 000, 13144A

UK: 999, 112

Restriction filter examples

Line and set restrictions are shown in Figure 135 and Figure 136.

In Figure 135, a caller using line 001 could only dial long-distance numbers to area codes 212 and 718. A caller using line 003 could not dial any long-distance numbers. A caller using line 005 could dial long-distance numbers to area codes 212, 718, and 415.

Tips: To restrict dialing from outside the system (once a caller gains remote access), apply restriction filters to incoming external lines (as remote restrictions).

Figure 135 Line restriction example

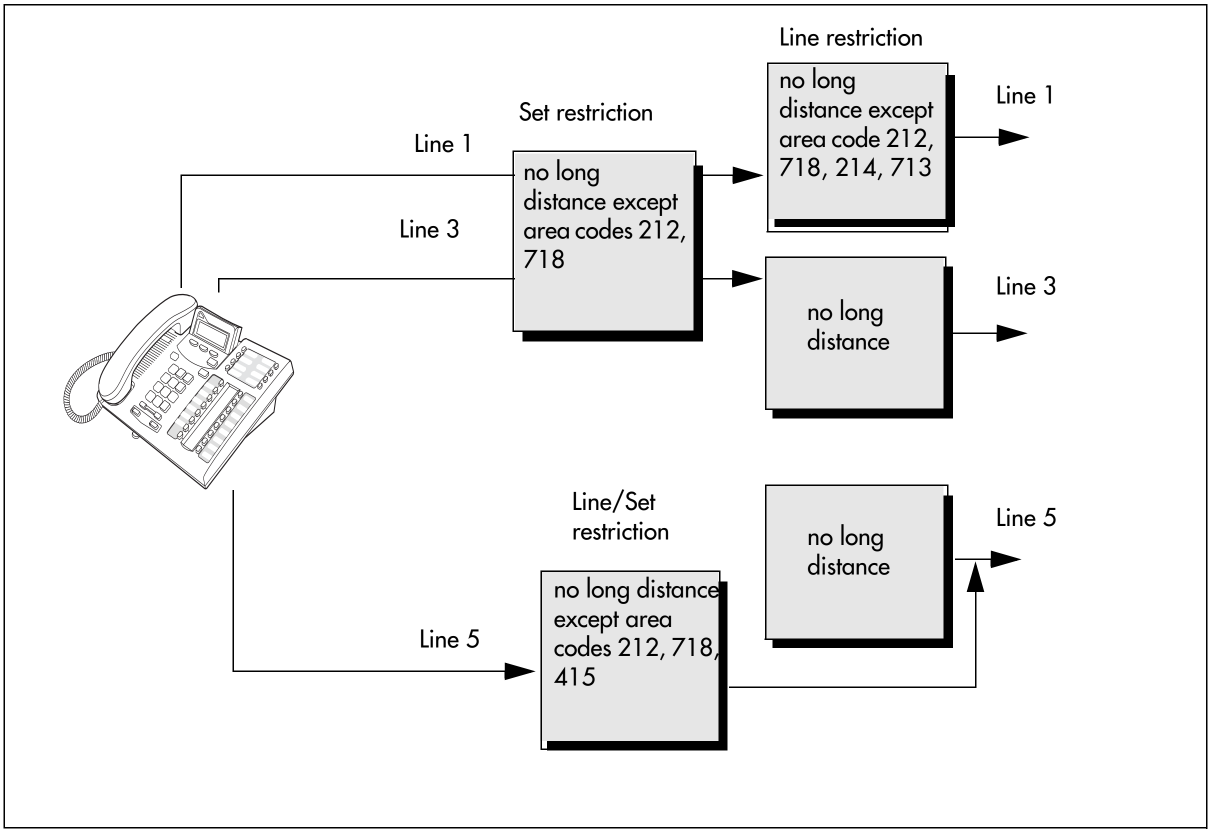
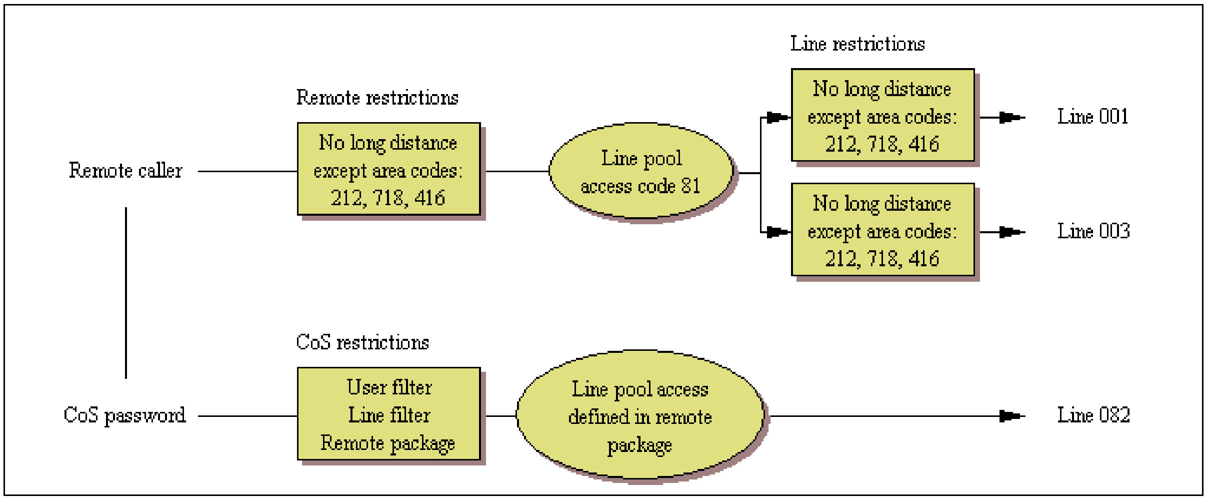

Line restriction
no long distance except area code 212, 718, 214, 713
Line 1
Set restriction
Line 1
no long distance except area codes 212, 718
Line 3
no long distance
Line 3
Line/Set restriction
no long distance
Line 5
no long distance except area codes 212, 718, 415
Line 5

Figure 136, dialed digits must pass both the remote restriction and the line restriction. A remote caller can override these filters by dialing the DISA DN and entering a CoS password.

Figure 136 Remote line restriction example


Remote caller
Remote restrictions
No long distance except area codes: 212, 718, 416
Line pool access code 81
Line restrictions
No long distance except area codes: 212, 718, 416
Line 001
No long distance except area codes: 212, 718, 416
Line 003
CoS password
CoS restrictions
User filter
Line filter
Remote package
Line pool access defined in remote package
Line 082

Remote call-in programming

There are three aspects to remote call ins:

- Setting up lines to allow users access to the system ("Creating Direct Inward System Access (DISA)" on page 445.
- Setting up Remote Access Packages that determine what services the remote users can access.
- Setting up CoS passwords for users calling in through the PSTN on lines programmed with DISA. ("Defining CoS passwords" on page 448)

Creating Direct Inward System Access (DISA)

To control access from the public or private network, you can configure auto-answer trunks to answer with DISA. Remote callers hear a stuttered dial tone and must then enter a CoS password that determines what they are allowed to do in the system.

- Auto-answer T1 loop start and T1 E&M trunks are configured to answer with DISA by default.
- T1 DID trunks: You cannot configure T1 DID trunks to answer with DISA. If you want incoming T1 DID calls to be answered with DISA, configure the system with a DISA DN. Incoming T1 DID calls that map onto the DISA DN are then routed to a line that has DISA.
- You cannot program a DISA DN or Auto DN to VoIP trunks, because they act as auto-answer lines for private networks. However, you still need to assign remote access packages to the VoIP trunks, to ensure that remote access restrictions are properly applied to incoming calls trying to access the system or the system network.

This section also includes information about:

- "Remote access line settings"
- "Remote access on loop start trunks" on page 446
- "Remote access on T1 DID and PRI trunks" on page 446

- “Remote access on DPNSS lines” on page 447
- “Remote access on a private network” on page 447

Remote access line settings

The remote access feature allows callers elsewhere on the private or the public network to access your BCM by dialing directly and not going through the attendant. After the remote user is in the system, they can use some of the system resources. You must enable remote access in programming before callers can use it.

BCM supports remote system access on a number of trunk types which may require the remote caller to enter a password for DISA.

The system resources, such as dialing capabilities, line pool access and feature access, that a remote user may access depends on the CoS password assigned to them. See “Defining CoS passwords” on page 448.

➡ **Note:** Callers remotely access the BCM remote features setting by pressing * and the appropriate page code. See the *BCM 4.0 Device Configuration Guide* (N0060600) for a list of feature codes.

Remote access on loop start trunks

Loop start trunks provide remote access to BCM from the public network. They must be configured to be auto-answer to provide remote system access.

A loop start trunk **must** have disconnect supervision if it is to operate in the auto-answer mode. T1 E&M trunks always operate in disconnect supervised mode.

When a caller dials into the system on a line that has auto-answer without DISA, the system answers with system dial tone and no CoS password is required. In this case, the remote access package assigned to the line controls system capabilities.

When a caller dials in on a line that has auto-answer with DISA, the system answers with stuttered dial tone. This is the prompt to enter a CoS password that determines which system capabilities are available to the caller.

Remote access on T1 DID and PRI trunks

Remote system access on T1 DID trunks is similar to that of T1 E&M trunks connected to a private network. The main differences are:

- A remote caller is on the public network dialing standard local or long distance telephone numbers.

- DISA cannot be administered to a T1 DID and PRI trunk. You can program the dialed digits to match those of a specific target line DN, the DISA DN or the Auto DN. If you program the dialed digits to the DISA DN, only the incoming calls that match the programmed DN will receive a DISA dial tone. Incoming calls with other digits will route to a target line.

Remote access on DPNSS lines

A remote caller can access a BCM system dial tone, select a line pool that contains exchange lines or DPNSS lines, then dial a number. The procedure is identical to dialing an outside number from an extension in the local system. The main features are:

- Calls coming from another switch to the BCM system are routed in two ways, depending on the Answer mode that you program. If the **Answer mode** is set to **Manual**, and the line is assigned to ring at an extension, the incoming call automatically rings at the assigned extension. If **Answer mode** is set to **Auto**, BCM automatically answers the incoming call. Because most other DPNSS features are extension-specific, Nortel recommends that all DPNSS lines are configured as auto-answer lines.
- The Page feature is available to both remote callers and callers within the system. A remote caller must have DTMF capability to access the Page feature.
- The line redirection feature allows the originating party to redirect a call that is waiting a connection or re-connection to an alternate destination after a time-out period. Failed calls can be redirected. Priority calls cannot be redirected.

Remote access on a private network

Systems connected to the private network deliver the last dialed digits to the destination BCM system for interpretation. The destination BCM system matches the digits to a target line or interprets the digits as a remote feature request. BCM then routes the call to the specified target line or activates the remote feature.

- By default, T1 E&M trunks are set to answer with DISA. For auto-answer T1 E&M trunks connected to a private network, change the default so that the trunks are **not** answered with DISA. If an auto-answer T1 E&M trunk is configured to answer with DISA, the system tries to interpret any received digits as a CoS password.
- The DISA DN and the Auto DN allow auto-answer private network and DID calls, in the same way that calls on auto-answer loop start and auto-answer T1 E&M trunks can be answered, with or without DISA. These DNs are described in “Understanding access codes” on page 259.
- Answer with DISA cannot be administered to a PRI trunk. Instead, you can program the dialed digits to match those of a specific target line DN, the DISA DN or the Auto DN on the other system.
- Answer with DISA cannot be administer to voice over IP (VoIP), since they do not connect systems outside the private network. However, a user calling in remotely on another system on the network can use the trunk to access the system or a user calling in on a PSTN line can use the trunk to access the private network. To provide control for this type of access, ensure that you specify remote access packages for the trunk.

Defining remote access packages

The Remote access packages setting allows you to control the remote access to line pools and remote page.

Create a remote access package by defining the system line pools remote users can access. You then assign the package to individual lines, and to a particular Class of Service password (see “Defining CoS passwords” on page 448).

Defining CoS passwords

CoS passwords permit controlled access to the system resources by both internal and remote users.

- When an internal user enters a CoS password at a telephone, the restriction filters associated with the CoS password apply instead of the normal restriction filters.
- Similarly, when a remote user enters a CoS password on an incoming auto-answer line, the restriction filters and remote package associated with their CoS password apply instead of the normal restriction filters and remote package.

Notes about CoS passwords

The CoS password can define the set of line pools that may be accessed and whether or not the user has access to the paging feature.

The class of service (CoS) that applies to an incoming remote access call is determined by:

- the filters that you apply to the incoming trunk
- the CoS password that the caller used to gain access to BCM.
- in cases where DISA is not automatically applied to incoming calls, the remote caller can change the class of service by dialing the DISA DN and entering a CoS password.

Remote users can access system lines, line pools, the Page feature, and remote administration. The exact facilities available to you through remote access vary depending on how your installer set up your system.

Note: If the loop start line used for remote access is not supervised, auto-answer does not function and the caller hears ringing instead of a stuttered tone or the system dial tone.

Security Note:
CoS password security and capacity

- Determine the CoS passwords for a system randomly and change them on a regular basis.
- Users should memorize their CoS passwords and keep them private. Typically, each user has a separate password. However, several users can share a password or one user can have several passwords.
- Delete individual CoS passwords or change group passwords when employees leave the company.
- A system can have a maximum of 100 six-digit CoS passwords (00 to 99).

To maintain the security of your system, the following practices are recommended:

- Warn a person to whom you give the remote access number to keep the number confidential.
- Change CoS passwords often.
- Warn a person to whom you give a CoS password, to memorize the password and not to write it down.
- Delete the CoS password of a person who leaves your company.

Security Note: Remote users can make long distance calls. Remember that a remote user can make long distance calls that are charged to your company. They can also access line pools and make page announcements in your office.

External access tones

You can hear some of the following tones when accessing BCM from a remote location. Table 105 shows the different types of tones and what they mean.

Table 105 External access tones (Sheet 1 of 2)

Tone	What it means
System dial tone	You can use the system without entering a CoS password.
Stuttered dial tone	Enter your CoS password.

Table 105 External access tones (Sheet 2 of 2)

Busy tone	You have dialed a busy line pool access code. You hear system dial tone again after five seconds.
Fast busy tone	You have done one of the following: • Entered an incorrect CoS password. Your call disconnects after five seconds. • Taken too long while entering a CoS password. Your call disconnects after five seconds. • Tried to use a line pool or feature not permitted by your Class of Service. You hear system dial tone again after five seconds. • Dialed a number in the system which does not exist. Your call disconnects after five seconds.
	IP trunk lines do not produce tones when accessed from a remote location.

Chapter 53

Call Security: Configuring Direct Inward System Access (DISA)

This following describes the telephony configuration that allows users to call from a remote site into the system to access system features.

The following paths indicate where to access DISA settings in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Resources > Telephony Resources; Configuration > Telephony > Dialing Plan > Public Network; Configuration > Telephony > Dialing Plan > Private Network**
- Telset interface: ****CONFIG > Hardware; System > Access codes**

Task: Configuring DISA DNs, Auto DNs, Answering with DISA
• Set up the system parameters for system users to call into the from a remote location. Note that Remote Access Packages are required for private network trunks, as well.

Refer to the following:

- "Remote access overview" on page 451
- "Setting up remote access on lines" on page 453

Remote access overview

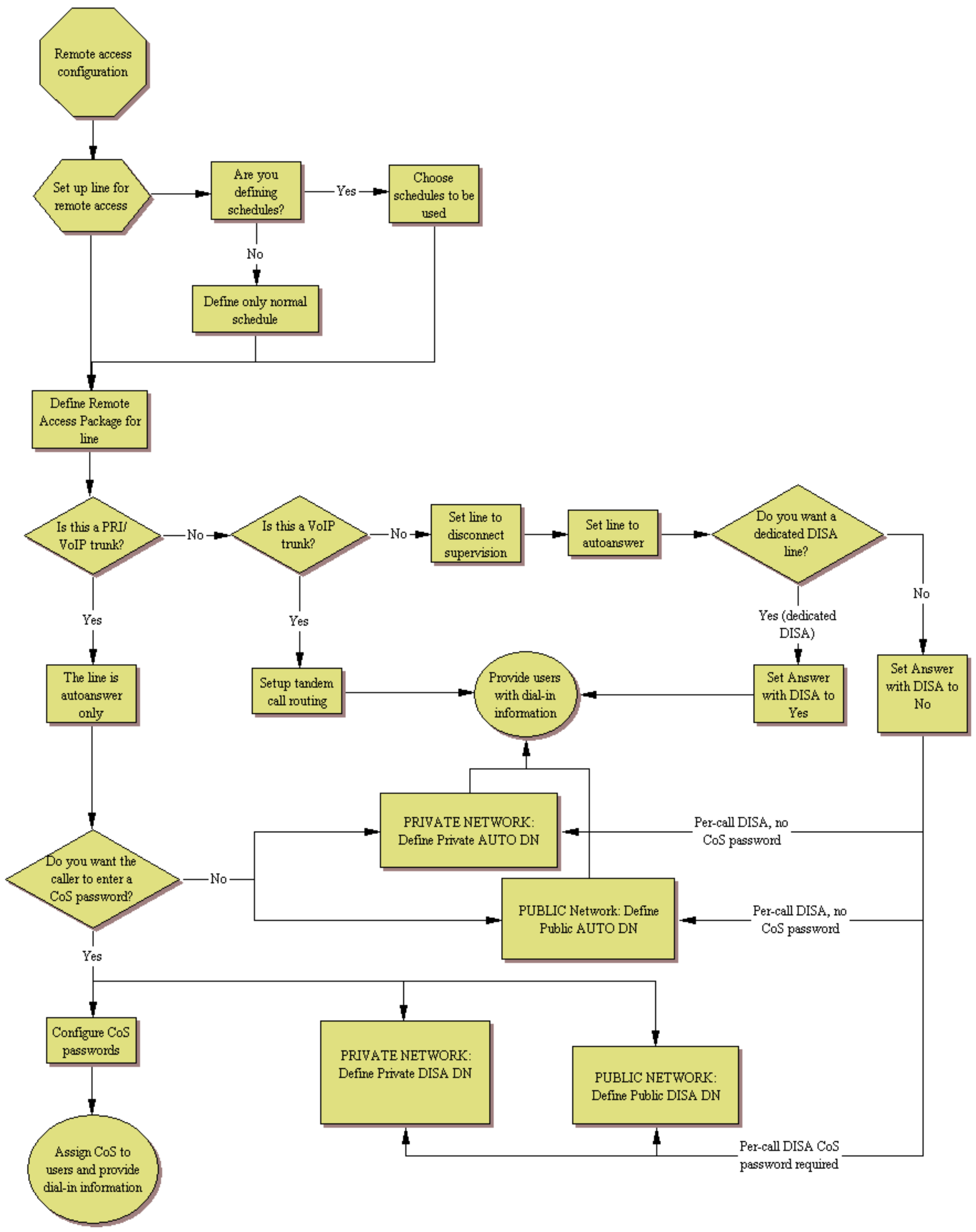
To control access from the public or private network, you can configure auto-answer trunks to answer with DISA. Remote callers hear a stuttered dial tone and must then enter a CoS password that determines what they are allowed to do in the system.

- Auto-answer T1 loop start and T1 E&M trunks are configured to answer with DISA by default.
- T1 DID trunks: You cannot configure T1 DID trunks to answer with DISA. If you want incoming T1 DID calls to be answered with DISA, configure the system with a DISA DN. Incoming T1 DID calls that map onto the DISA DN are then routed to a line that has DISA.
- You cannot program a DISA DN or Auto DN to VoIP trunks, because they act as auto-answer lines for private networks. However, you still need to assign remote access packages to the VoIP trunks, to ensure that remote access restrictions are properly applied to incoming calls trying to access the system or the system network.

For specific line programming, refer to the sections under "Setting up remote access on lines" on page 453.

Figure 137 provides an overview of the remote access configuration process.

Figure 137 Remote access task overview


Remote access configuration
Set up line for remote access
Are you defining schedules?
Yes
Choose schedules to be used
No
Define only normal schedule
Define Remote Access Package for line
Is this a PRI/ VoIP trunk?
No
Is this a VoIP trunk?
No
Set line to disconnect supervision
Set line to autoanswer
Do you want a dedicated DISA line?
No
Yes
Yes
Yes (dedicated DISA)
The line is autoanswer only
Setup tandem call routing
Provide users with dial-in information
Set Answer with DISA to Yes
Set Answer with DISA to No
PRIVATE NETWORK: Define Private AUTO DN
Per-call DISA, no CoS password
Do you want the caller to enter a CoS password?
No
PUBLIC Network: Define Public AUTO DN
Per-call DISA, no CoS password
Yes
Configure CoS passwords
PRIVATE NETWORK: Define Private DISA DN
PUBLIC NETWORK: Define Public DISA DN
Per-call DISA CoS password required
Assign CoS to users and provide dial-in information

Setting up remote access on lines

Setting up remote access on different types of trunks requires you to understand the trunk properties and how you want the system to answer the dial-in calls.

Refer to the following topics:

- "Remote access on loop start trunks" on page 453
- "Remote access on T1 DID trunks" on page 453
- "Remote access on PRI" on page 454
- "Remote access on DPNSS lines" on page 454
- "Remote access on a private network" on page 455
- "Other programming" on page 455

Remote access on loop start trunks

Loop start trunks provide remote access to BCM from the public network. They must be configured to be auto-answer to provide remote system access. Refer to "Configuring lines: T1-Loop start" on page 187.

A loop start trunk **must** have disconnect supervision if it is to operate in the auto-answer mode. T1 E&M trunks always operate in disconnect supervised mode.

When a caller dials into the system on a line that has auto-answer without DISA, the system answers with system dial tone and no CoS password is required. In this case, the restriction filters assigned to the line control system capabilities available to the caller.

When a caller dials in on a line that has auto-answer with DISA, the system answers with stuttered dial tone. This is the prompt to enter a CoS password that determines which system capabilities are available to the caller.

Remote access on T1 DID trunks

Remote system access on T1 DID trunks is similar to that of T1 E&M trunks connected to a private network.

The main differences are:

- A remote caller is on the public network dialing standard local or long distance telephone numbers.
- The digits received are delivered by the central office.
- DISA cannot be administered to a T1 DID trunk. You can program the dialed digits to match those of a specific target line DN, the DISA DN or the Auto DN. If you program the dialed digits to the DISA DN, only the incoming calls that match the programmed DN will receive a DISA dial tone. Incoming calls with other digits will route to a target line.

Refer to "Configuring lines: T1-E&M" on page 181, "Configuring lines: T1-DID" on page 199.

Remote access on PRI

Remote system access on PRI trunks is similar to that of T1 E&M trunks connected to a private network.

The main differences are:

- A remote caller is on the public network dialing standard local or long-distance telephone numbers.
- The digits received are delivered by the central office.
- Answer with DISA cannot be administered to a PRI trunk. Instead, you can program the dialed digits to match those of a specific target line DN, the DISA DN or the Auto DN.
- North America: Use incoming Call by Call (CbC) Service routing to map the call type to the DISA DN.

  With FX, INWATS, 900, and SDS service types, either a Service Id (SID) or a CDN is mapped to Target Line Receive Digits. This is programmed under "Configuring PRI Call-by-Call services" on page 174. DISA may be accessed by having the SID or CDN map to the DISA DN. This example has a Receive Digit Length = 4, DISA DN = 1234, and CbC Routing with (Service Type = FX, Map from SID = 2, Map to digits = 1234).

  A call presented to the BCM system with service type FX and SID 2 will be handled as follows:

  — The ISDN setup message will specify FX with SID = 2

  — The FX SID = 2 will be mapped to DISA DN digits 1234

  — The call will be answered with DISA.

Refer to "Configuring lines: PRI" on page 171.

Remote access on DPNSS lines

A remote caller can access a BCM system dial tone, select a line pool that contains exchange lines or DPNSS lines, then dial a number. The procedure is identical to dialing an outside number from an extension in the local system. The main features are:

- Calls coming from another switch to the BCM system are routed in two ways, depending on the Answer mode that you program. If the **Answer mode** is set to **Manual**, and the line is assigned to ring at an extension, the incoming call automatically rings at the assigned extension. If **Answer mode** is set to **Auto**, BCM automatically answers the incoming call. Because most other DPNSS features are extension-specific, Nortel recommends that all DPNSS lines are configured as auto-answer lines.
- The Page feature is available to both remote callers and callers within the system. A remote caller must have DTMF capability to access the Page feature.
- The line redirection feature allows the originating party to redirect a call that is waiting a connection or re-connection to an alternate destination after a time-out period. Failed calls can be redirected. Priority calls cannot be redirected.

Refer to "Private networking: DPNSS network services (UK only)" on page 365.

Remote access on a private network

Systems connected to the private network deliver the last dialed digits to the destination BCM system for interpretation. The destination BCM system matches the digits to a target line or interprets the digits as a remote feature request. BCM then routes the call to the specified target line or activates the remote feature.

- By default, T1 E&M trunks are set to answer with DISA. For auto-answer T1 E&M trunks connected to a private network, change the default so that the trunks are **not** answered with DISA. If an auto-answer T1 E&M trunk is configured to answer with DISA, the system tries to interpret any received digits as a CoS password.
- The DISA DN and the Auto DN allow auto-answer private network and DID calls, in the same way that calls on auto-answer loop start and auto-answer T1 E&M trunks can be answered, with or without DISA. These DNs are described in "Dialing plan: Private network settings" on page 317.
- Answer with DISA cannot be administered to a PRI trunk. Instead, you can program the dialed digits to match those of a specific target line DN, the DISA DN or the Auto DN on the other system.
- Answer with DISA cannot be administer to voice over IP (VoIP), since they do not connect systems outside the private network. However, a user calling in remotely on another system on the network can use the trunk to access the system or a user calling in on a PSTN line can use the trunk to access the private network. To provide control for this type of access, ensure that you specify remote access packages for the trunk. This type of call is called a tandem call.

Other programming

- "Call security: Remote access packages" on page 463
- "Configuring CoS passwords for remote access" on page 467

Chapter 54
Call security: Restriction filters

The following describes the panels that are used to enter restriction filters and restriction overrides. You can have a maximum of 100 restriction filters on the system.

The following paths indicate where to access restriction filter settings in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Call Security > Restriction Filters**
- Telset Interface: ****CONFIG > Terminals and Sets, or **CONFIG > Lines**

Click one of the following links to connect with the type of information you want to view:

Panels	Tasks	Feature notes
"Restriction filters" on page 457	"Adding a restriction filter and exceptions" on page 459	"Default filters" on page 460
Using restriction filters:	"Restrictions (Line and Remote)" on page 156 "Class of Service table" on page 467 "Hospitality - General" in the *BCM 4.0 Device Configuration Guide* (N0060600)	

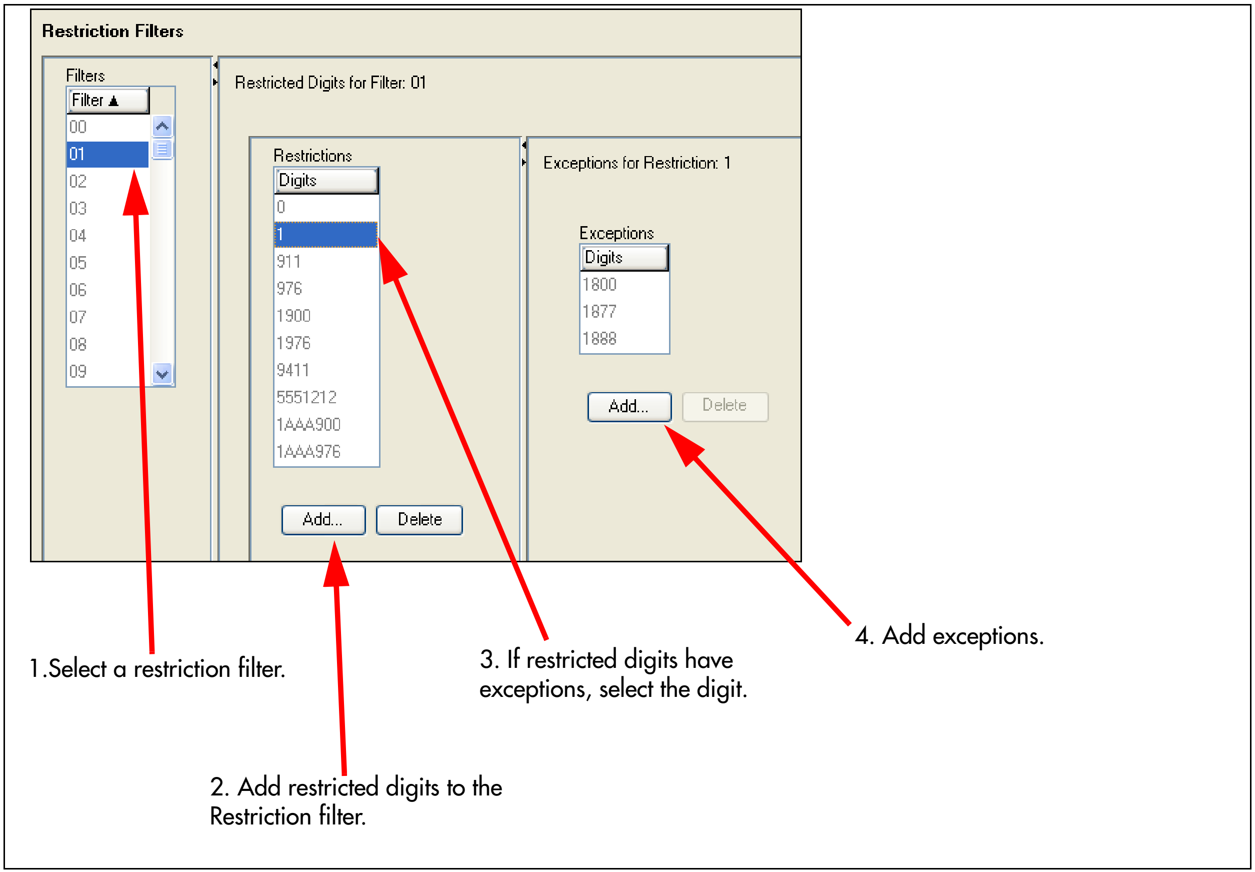
Click the navigation tree heading to access general information about restriction filters.

Restriction filters

Restrictions are used to restrict outbound dialing. For example, restrictions can be applied to restrict dialing 1-900 numbers.

The restriction filters panel contains three list boxes. You progress from left to right as you populate the information.

Figure 138 Restriction Filters panels



1.Select a restriction filter.

2. Add restricted digits to the Restriction filter.

3. If restricted digits have exceptions, select the digit.

4. Add exceptions.

Table 106 provides a description of the fields on the Restriction filters panel.

Table 106 Restriction filters and exceptions fields (Sheet 1 of 2)

Attribute	Value	Description
Filters table		
Filter	<00-99>	This is the list number for the filter. This is the number that you will use on the configuration panels that require restriction filter entries.
Restrictions table		
Digits	<dialstring digit(s)>	For each filter, enter the restriction digit dial string, based on what the restriction is for. The dial string is the number that is restricted from being dialed on the system. Also refer to "Default filters" on page 460. Note: The wildcard *A* (Any) can be used as part of the dialstring.
Actions:		
Add	Refer to "Adding a restriction filter and exceptions" on page 459.	
Delete	1. On the Filters table, select the filter where you want to delete information. 2. On the Restrictions table, select one or more restrictions to delete. 3. Click **Delete**. 4. Click **OK**.	

Table 106 Restriction filters and exceptions fields (Sheet 2 of 2)

Attribute	Value	Description
Exceptions table		
Digits	<dialstring digit(s)>	For each restriction digit, enter any numbers that should dial out, despite the restriction. **Note:** The wildcard *A* (Any) can be used as part of the dialstring.
Actions:		
Add	Refer to "Adding a restriction filter and exceptions" on page 459	
Delete	1. On the Filters table, select the filter where you want to delete information. 2. On the Restrictions table, select the restriction filter that has the exception that you want to delete. 3. On the Exceptions table, click one or more of the exceptions. 4. Under the Exceptions table, click **Delete**. 5. Click **OK**.	

The default values for restriction filters are based on country profile. Refer to "Default filters" on page 460 and "Default filters for other common profiles" on page 461.

Adding a restriction filter and exceptions

To add a restriction filter

1. Click **Configuration** > **Telephony** > **Call Security** > **Restriction filters**.
2. Click the Filter to add the restriction.
   The **Restrictions** panel appears.
3. Click **Add.**
   The **Add Restriction** dialog box appears.
4. Enter the digits that you want to restrict if they precede a dial string going out of the system.
5. Click **OK**.
6. Repeat steps 3 and 4 for all filters you want to add.
7. If you need to apply overrides to a filter, on the Restricted table, click the restricted digit to which you want to add overrides.
8. Under the Exceptions panel, click **Add**.
   The **Add Exception** dialog box appears.
9. Enter the number that you want to allow when this restriction is in effect.
10. Repeat steps 7 and 8 for all overrides you want to add to this filter.
11. Repeat steps 6 to 9 for all the filters to which you want to add overrides.
12. Click **OK**.

13 Next steps: Assign filters to lines, DN records and class of service (CoS) passwords for remote access.

Default filters

The following provides a list of the default restriction filters for North America and other common profiles:

- “Default filters for the North America profile” on page 460
- “Default filters for other common profiles” on page 461

Default filters for the North America profile

Filter 00 permits unrestricted dialing and cannot be changed.

Filter 01 is pre-programmed with 10 restrictions and some associated overrides. In Filter 01, Restriction 02 and Override 001 allow long distance toll free calls.

The dialing string 911, which is the number for emergency assistance in North America, is included as both a restriction and an override in Filter 01. This arrangement prevents anyone from blocking calls for emergency assistance on lines or sets using the default filter.

Table 107 Default restriction filters

Filter	Restrictions (denied)	Overrides
00	Unrestricted dialing	
01	01: 0	
	02: 1	001: 1800 002: 1877 003: 1888
	03: 911	001: 911
	04: 411	
	05: 976	
	06: 1976	
	07: 1AAA976	
01	08: 1900	
	09: 1AAA900	
	10: 5551212	
02 - 99	No restrictions or exceptions programmed	

Note: Default filters are loaded when the system is initialized. A cold start restores the default filters.

Filters 02, 03, and 04, although not preset with restrictions and overrides, are the default filters in these programming headings:

Filter	Heading	Sub-heading
02	System DNs	Set restrictions
03	Lines	Line restriction
04	Lines	Remote restriction

Default filters for other common profiles

Three profiles have global overrides which do not appear in Element Manager restriction programming and cannot be changed.

Australia: 000, 13144A

UK: 999, 112

Chapter 55

Call security: Remote access packages

The following describes the telephony configuration that is used to control access to system lines by calls coming in from outside the system.The remote access package also allows remote paging capabilities.

➡ **Note:** Callers dialing into the system over private network lines are also considered remote callers.

The following paths indicate where to access remote access packages in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Call Security > Remote Access Packages**
- Telset interface: ****CONFIG > System prgrming > Remote Access**

This is a two-table panel, where you select a Remote Access Package number on the first panel and then add or delete the line pools from the second table.

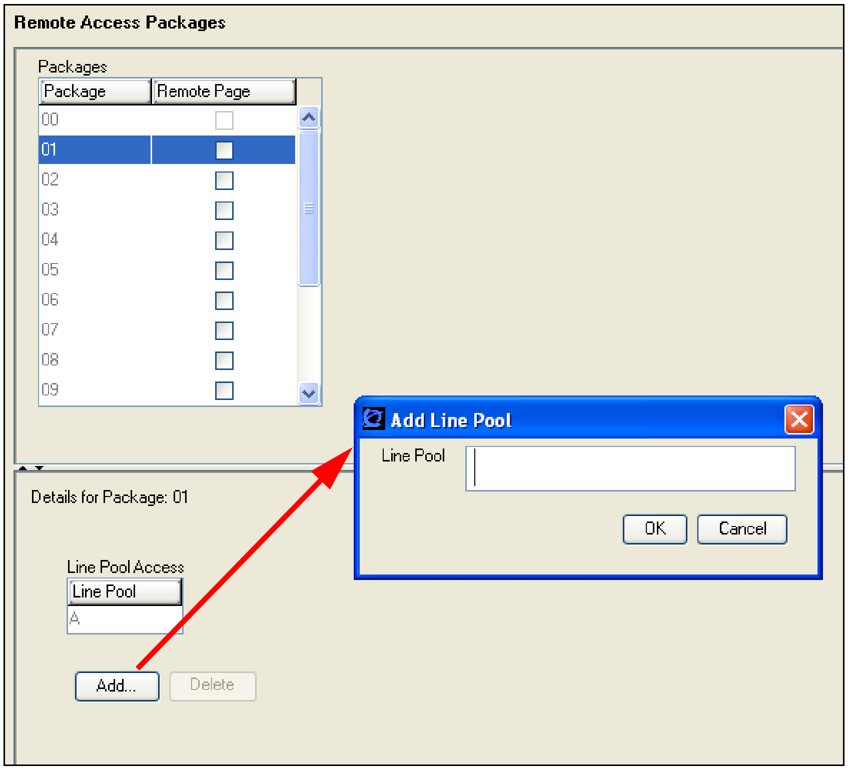
Panels/Subpanels	**Tasks**
"Configuring remote access packages" on page 463	"Restrictions (Line and Remote)" on page 156 (lines)
Also refer to:	"Call Security: Configuring Direct Inward System Access (DISA)" on page 451
	"Configuring CoS passwords for remote access" on page 467
Click the navigation tree heading to access general information about Hospitality services.	

Configuring remote access packages

Use these panels to add allowed line pools to up to 99 remote access packages.

Remote access packages are assigned to lines and class of service (CoS) passwords. Lines used for private networking need remote access packages because calls coming from other nodes on the network are considered remote call-ins by your system.

Figure 139 Remote Access Packages tables



Remote Access Packages

Packages

Package	Remote Page
00	□
01	□
02	□
03	□
04	□
05	□
06	□
07	□
08	□
09	□

Details for Package: 01

Line Pool Access

Line Pool
A

Table 108 describes each field on this panel.

Table 108 Remote Access Packages (Sheet 1 of 2)

Attribute	**Values**	**Description**
Packages table		
Package	<00-99>	This designates the package number. This is what is entered in the fields for lines programming for remote access.
Remote Page	<check box>	Select check box if you wish to allow remote callers access to paging. Note: **Remote paging is not supported on IP trunks.**
Line Pool Access table		
Line pool	<A to O> (digital lines and VoIP lines) BlocA to F (PRI and ETSI QSIG lines)	Choose the line pool for which you want this package to be available.
Actions		

Table 108 Remote Access Packages (Sheet 2 of 2)

Add (line pool)	Package 00 is the default package and cannot be deleted. It provides no access to any line pools. 1. On the Packages table, select the remote package number that you want to configure. 2. Under the Line Pool Access table, click **Add**. 3. In the Add dialog, enter a line pool. 4. Click **OK** to save the pool. 5. Next steps: Add remote access packages to lines and CoS passwords.
Delete (line pool)	1. On the Packages table, select the remote package number where you want to delete line pools. 2. On the Line Pool Access table select one or more line pools to delete. 3. Click **Delete**. 4. Click **OK**.

The following is an example of how a remote access package works.

- Inbound PRI calls are on line pool BlocA
- Outbound calls are on analog lines using Pool A

If users coming in on the PRI are to be able to access outbound trunks on Pool A then the lines in BlocA must be in a remote package that allows access to Pool A

Chapter 56
Configuring CoS passwords for remote access

The Class of Service panel allows you to configure passwords for system users who will be dialing into the system over a PSTN/private network to use system features, or for users who must bypass local restrictions on telephones.

The following paths indicate where to access the Class of Service settings in Element Manager and through Telset Administration:

- Element Manager: **Configuration > Telephony > Call Security > Class of Service**
- Telset interface: ****CONFIG > Passwords**

Click one of the following links to connect with the type of information you want to view:

Panel tabs	Tasks/Features
"Class of Service table" on page 467	"External access tones" on page 471
Also refer to:	"Call security: Restriction filters" on page 457 "Call Security: Configuring Direct Inward System Access (DISA)" on page 451 "Call security: Remote access packages" on page 463

Click the navigation tree heading to access general information about user management.

CoS passwords permit controlled access to the system resources by both internal and remote users.

- When an internal user enters a CoS password at a telephone, the restriction filters associated with the CoS password apply instead of the normal restriction filters.
- Similarly, when a remote user enters a CoS password on an incoming auto-answer line, the restriction filters and remote package associated with their CoS password apply instead of the normal restriction filters and remote package.

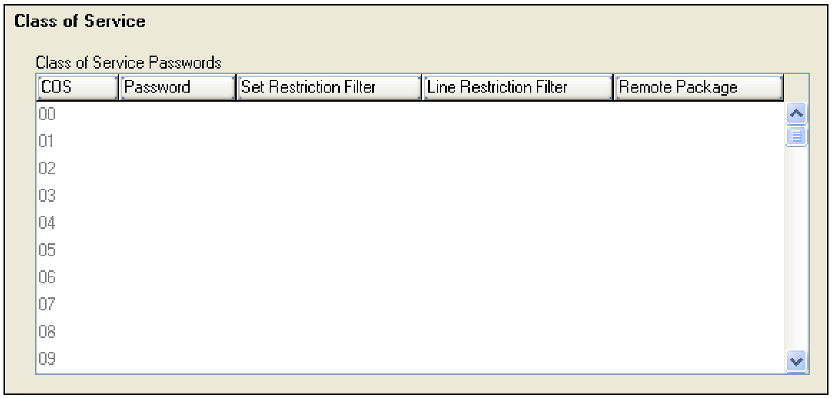
Class of Service table

Refer to the following information:

- "Notes about CoS passwords" on page 469
- "External access tones" on page 471

Security Note: Change passwords frequently to discourage unauthorized access.

Figure 140 Class of Service table panel



COS	Password	Set Restriction Filter	Line Restriction Filter	Remote Package
00				
01				
02				
03				
04				
05				
06				
07				
08				
09				

Table 109 describes the fields on this panel.

Table 109 CoS password values

Attribute	**Values**	**Description**
CoS	<CoS 00- CoS 99> Read-only	These numbers identify the password position to the system.
Password	<six digits>	Enter a combination of numbers that the user needs to dial to get into the system. Refer to "Notes about CoS passwords" on page 469.
Set Restriction Filter	None Filter <plus a two-digit user filter>	Assign a restriction filter to a Class of Service password. The user filter associated with the Class of Service password replaces any normally-applicable set restriction, line/set restriction, and remote restriction. The default setting (**None**), means that any normally- applicable filters (set restriction, line/set restriction, or remote restriction) still apply.
Line Restriction Filter	None Filter <plus a two-digit line filter>	Assign a specific line restriction to a Class of Service password. The line filter associated with the Class of Service password replaces any normally applicable line restriction. The default setting (**None**), means that any normally applicable line filter still applies.
Remote Package	None Package <plus a two-digit remote package>	Refer to "Call security: Remote access packages" on page 463 for more information.

Adding or modifying a CoS password values

Programming references:

- "Notes about CoS passwords" on page 469
- "External access tones" on page 471

Note: You can add a maximum of 99 CoS Passwords.

To add or modify a CoS password

1. On the Class of Service table, click the CoS line to which you want to add or modify a password.
2. Select the field you want to change and enter the appropriate information:
   - Name: Enter a descriptive name for the password or user
   - Password: Enter a set of six digits that are unique from any other CoS password
   - Set Restriction Filter: If you want the user to be able to override set and line/set restrictions for the number being called, enter the allowed filters.
   - Line Restriction Filter: If you want the user to be able to override the line restrictions that the call uses to access the system, enter the allowed filters here.
   - Remote Package: Enter the remote package that you want the system to use to determine the level of access the user will have to system features.

Notes about CoS passwords

The CoS password can define the set of line pools that may be accessed and whether or not the user has access to the paging feature. The password all defines which restrictions are applied.

The class of service (CoS) that applies to an incoming remote access call is determined by:

- the filters that you apply to the incoming trunk
- the CoS password that the caller used to gain access to BCM.
- in cases where DISA is not automatically applied to incoming calls, the remote caller can change the class of service by dialing the DISA DN and entering a CoS password.

Remote users can access system lines, line pools, the Page feature, and remote administration. The exact facilities available to you through remote access vary depending on how your installer set up your system.

Note: Remote paging is not available on IP trunks.

Security Note:
CoS password security and capacity

- Determine the CoS passwords for a system randomly and change them on a regular basis.
- Users should memorize their CoS passwords and keep them private. Typically, each user has a separate password. However, several users can share a password or one user can have several passwords.
- Delete individual CoS passwords or change group passwords when employees leave the company.
- A system can have a maximum of 100 six-digit CoS passwords (00 to 99). CoS passwords must be unique.

To maintain the security of your system, the following practices are recommended:

- Warn a person to whom you give the remote access number to keep the number confidential.
- Change CoS passwords often.
- Warn a person to whom you give a CoS password, to memorize the password and not to write it down.
- Delete the CoS password of a person who leaves your company.

Security note: Remote users can make long distance calls.
Remember that a remote user can make long distance calls that are charged to your company. They can also access line pools and make page announcements in your office.

CoS examples

Example: Using the CoS feature to access a restricted line.

A sales representative out of the office needs to make long distance calls to the European office. Your system has a leased line to Europe with reduced transatlantic charges. You provide the sales representative with a Class of Service password that gives access to the transatlantic line. The sales representative can telephone into the system (DISA DN) from a hotel, enter the Class of Service password, and then use a destination code to access the leased transatlantic line to make calls.

To access the system over a public network

1. Dial the system remote access number.
2. When you hear a stuttered dial tone, enter your CoS password.
3. Wait for the system dial tone.

To bypass the restriction filters on a telephone

1. Press **FEATURE 68**.
2. Enter the six-digit CoS password that allows the required type of call.
3. Enter the number to be dialed.

Example: Remote access over the public network bypassing the restrictions on a telephone

To use the system at a distance, you must use a telephone with tone dialing to call the system. Remote access is possible only on lines that your installer programs to auto-answer calls.

To use paging on a remote system, press * followed by the feature code. When you are calling from within BCM, press * instead of **FEATURE**.

In some conditions, you can experience lower volume levels when using the system from a distance.

External access tones

You can hear some of the following tones when accessing BCM from a distance. Table 110 shows the different types of tone and what they mean.

Table 110 External access tones

Tone	What it means
System dial tone	You can use the system without entering a CoS password.
Stuttered dial tone	Enter your CoS password.
Busy tone	You have dialed a busy line pool access code. You hear system dial tone again after 5 seconds.
Fast busy tone	You have done one of the following: • Entered an incorrect CoS password. Your call disconnects after five seconds. • Taken too long while entering a CoS password. Your call disconnects after five seconds. • Tried to use a line pool or feature not permitted by your Class of Service. You hear system dial tone again after five seconds. • Dialed a number in the system which does not exist. Your call disconnects after five seconds.
	IP trunk lines do not produce tones when accessed from a remote location.

Chapter 57
Data networking overview

The BCM is a converged voice product, and can be connected to virtually any data network, to provide Voice over Internet Protocol (VoIP) support in either a Local Area Network (LAN) or Wide Area Network (WAN) environment. The BCM is also available with an integrated Broadband Ethernet or ADSL Router, which is intended to provide basic data networking and services, as well as Virtual Private Network (VPN) connectivity for small sites. Refer to “Virtual Private Networks (VPN)” on page 641 fore more information. With the router, the BCM can handle all data networking needs, including both VoIP and basic IP networking. The BCM is also available without a router, to provide VoIP capabilities to networks that already have an existing IP network.

What is data networking?

On the BCM, data networking refers to both standard IP data networks, as well as VoIP. These two types of networks are closely intertwined, and connect a wide range of IP devices - including IP telephones and computers - with the BCM and with external networks. The BCM with router can also handle all routing requirements.

For an more information about setting up networks “System telephony networking overview” on page 39.

About the BCM VoIP capability

The BCM provides VoIP functionality both within a LAN (Local Area Network), and across a WAN. It can contain IP telephones, which act similar to a traditional phone, but send their signals across data networks in the form of IP packets. The BCM can also contain IP trunks which connect offices together across an IP network.

For more information about VoIP refer to “VoIP overview” on page 391.

Network routing

The BCM is available with and without an internal router. With the router, it can handle all external connections necessary for a data-network, as well as control security on these connections. The standalone version of the BCM does not handle routing, but is suitable for IP networks where a router is already in place.

Configuring the BCM with data networks

To configure the BCM to work with a data network, go through each of the following steps:

- Complete the pre-installation checklist. This will make sure that you've made all necessary preparations for connecting the BCM. For information on completing the pre-installation checklist, "Prerequisites checklist" on page 475.
- Configure your router. If you already have a router already on your system, you must make some modifications to its configuration for use with the BCM.
- Configure IP settings on the BCM. For information on configuring IP settings on the BCM, "Configuring LAN resources" on page 489.
- Configure DHCP on the BCM. For information on configuring DHCP on the BCM, "Configuring DHCP" on page 583.

Chapter 58
Prerequisites checklist

Before you set up voice over IP (VoIP) trunks or IP telephones on a BCM, complete the following checklists to ensure that the system is correctly set up for IP telephony. Some questions do not apply to all installations.

This guide contains informing on various aspects of IP networking directly related to IP telephony functions. Refer to the *BCM 4.0 Device Configuration Guide* (N0060600) for specific information about configuring the data portion of the BCM.

Complete the following checklists:

- "Network diagram" on page 475
- "Network devices" on page 476
- "Network assessment" on page 477
- "Resource assessment" on page 477
- "Keycodes" on page 478
- "System configuration for IP functions" on page 478
- "Finding the published IP address" on page 479
- "Media gateway parameters for IP service" on page 482
- "VoIP trunks" on page 484
- "IP telephone records" on page 484

Network diagram

To aid in installation, a Network Diagram provides a basic understanding of how the network is configured. Before you install IP functionality, create a network diagram that captures all of the information described in the following table. If you are configuring IP telephones but not voice over IP (VoIP) trunks, you do not need to answer the last questions 1.d or 1.e.

Table 111 Network diagram prerequisites (Sheet 1 of 2)

Prerequisites	Yes
1.a Has a network diagram been developed?	
1.b Does the network diagram contain any routers, switches or bridges with corresponding IP addresses and bandwidth values for WAN or LAN links?	
1.c Does the network diagram contain IP Addresses, netmasks, and network locations of all BCMs?	
1.d Answer this if your system will use IP trunks, otherwise, leave it blank: Does the network diagram contain IP Addresses and netmasks of any other VoIP gateways that you need to connect to?	

Table 111 Network diagram prerequisites (Sheet 2 of 2)

Prerequisites	Yes
1.e Answer this only if your system will use a gatekeeper, otherwise, leave it blank: Does the network diagram contain the IP address for any Gatekeeper that may be used? **Note:** If the network has a Meridian 1 running IPT software, you cannot use a RadVision gatekeeper.	

Network devices

The following table contains questions about devices on the network such as firewalls, NAT devices, and DHCP servers.

- If the network uses public IP addresses, complete 2.d.
- If the network uses private IP addresses, complete 2.e. to 2.f.

Table 112 Network device checklist

Prerequisites	Yes	No
2.a Is the network using DHCP?		
2.b If so, are you using the DHCP server on the BCM? WLAN wireless IP telephones use either full DHCP or static IP addresses.		
2.c Is the network using private IP addresses?		
2.d Are there enough public IP addresses to accommodate all IP telephones and the BCM?		
2.e Does the system have a firewall/NAT device, or will the BCM be used as a firewall/NAT device?		
2.f If the BCM is to be used as a firewall/NAT device, do the firewall rules fit within the rules that the BCM provides?		

Network assessment

The following table questions are meant to ensure that the network is capable of handling IP telephony, and that existing network services are not adversely affected.

Table 113 Network assessment

Prerequisites	Yes	No
3.a Has a network assessment been completed?		
3.b Has the number of used and available switch ports in the LAN infrastructure been calculated?		
3.c Does the switch use VLANs? If so, get the VLAN port number and ID.		
3.d Have the used, and available, IP addresses for each LAN segment been calculated?		
3.e Has DHCP usage and location been recorded?		
3.f Has the speed and configuration of the LAN been calculated?		
3.g Have the estimated latency values between network locations been calculated?		
3.h Have the Bandwidth/CIR utilization values for all WAN links been calculated?		
3.i Has the quality of service availability on the network been calculated?		

Resource assessment

Answer the questions in the following table to determine if you have allocated sufficient resources on the BCM for IP telephony.

For information about changing the DS30 split for the Business Communications Manager and allocating media resources, refer to the *"Determining the resources you require" on page 86* (data sections).

Hardware restriction: IP telephony, including T.38 fax, cannot operate on a system that has PEC Is installed on the MSC. Your system must have PEC IIIs.

Table 114 Resource assessment

Prerequisites	Yes	No
4.a Has an analysis been done to determine which DS30 split is appropriate for the system? Has the DS30 split been changed to 3/5, if necessary?		
4.b Have all necessary media resources for IP trunks, clients, vmail, IP music, or WAN dialup been assigned or dedicated?		

Keycodes

All elements of VoIP trunks and IP telephony are locked by the BCM keycode system. Answer the questions in the following table to ensure you have the appropriate keycodes. You can purchase keycodes for the amount of access you want for your system. Additional keycodes can be added later, provided there are adequate resources to handle them. For information on determining the number of keycodes required, see the *Keycode Installation Guide* (N0060625).

Table 115 Keycodes

Prerequisites	Yes	No
5.a Complete this question only if you are using VoIP trunks: Do you have enough VoIP keycodes? Both H.323 trunks and SIP trunks use VoIP keycodes.		
5.b Complete this question only if you are using IP telephones: Do you have enough IP client keycodes? (Note: IP clients and IP telephones are a 1:1 ratio. Include any WLAN wireless IP telephones to your calculations. As soon as an IP telephone is registered, it occupies an IP client, whether it is active or not.).		
5.c If you are using VoIP trunks, do you need to activate MCDN features? **Note:** If MCDN is already configured on your system for private networking over PSTN lines, you do not need a separate MCDN keycode for VoIP trunks. SIP trunks do not support the MCDN protocol.		

System configuration for IP functions

Several sections of the BCM must be properly configured prior to IP telephony activation. Answer the questions in the following table to determine if your BCM has been correctly configured.

Table 116 BCM system configuration

Prerequisites	Yes	No
6.a Is the LAN functioning correctly with the BCM? You can test this by pinging other addresses around the network from the BCM.		
6.b Is the WAN functioning correctly with the BCM?		
6.c Have you determined the published IP address for the system? Refer to "Finding the published IP address" on page 479.		
6.d Have the necessary media gateway, IP client, and IP trunks resources been set? (Refer to "Media gateway parameters for IP service" on page 482.)		
6.e Has a dialing plan been created, taking into account special considerations for IP telephony and private and public networking?		

Finding the published IP address

The published IP address is the IP address used by computers on the public network to find the Business Communications Manager. For example, if a Business Communications Manager has a LAN interface (LAN1) that is connected only to local office IP terminals and a WAN interface (WAN1) that is connected to the public network, then WAN1 should be set to the published IP address.

Setting the Global IP (published IP)

To set the published IP address

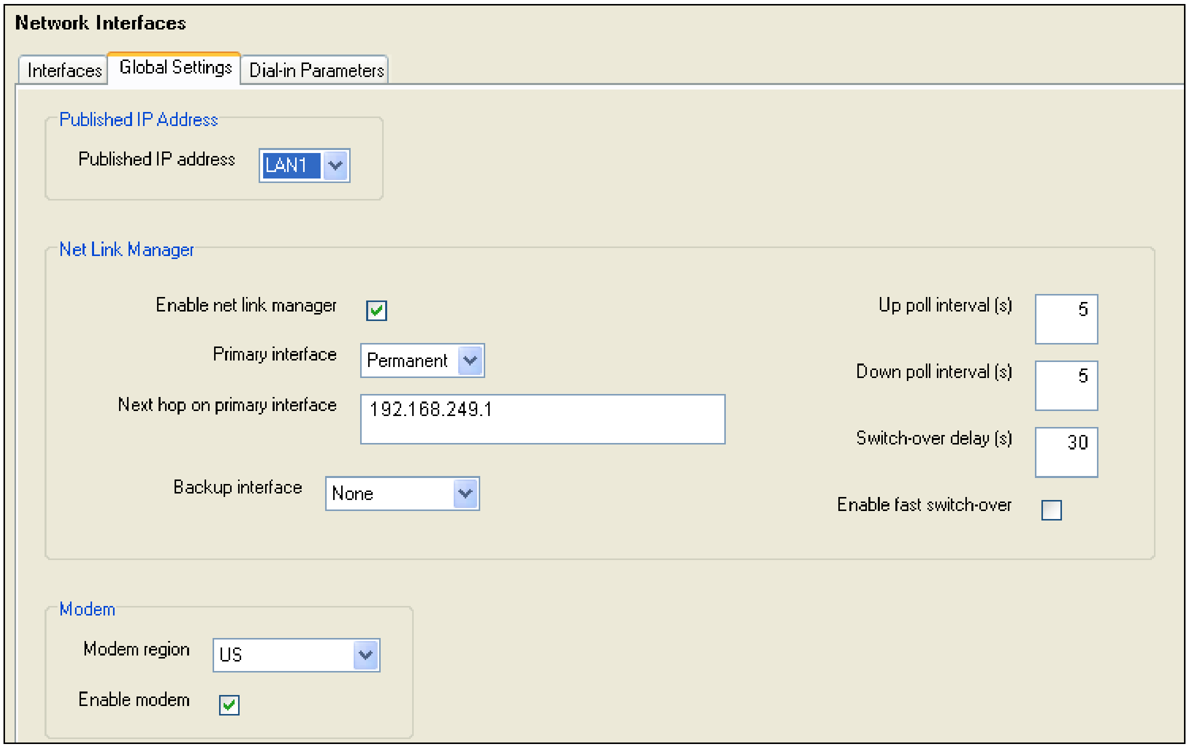
1 Click **Configuration > Resources > Network Interfaces**.

2 In the interfaces tab, select the device to configure the IP address.
The details panel for that device appears.

3 Click the Global Settings Tab.
The Global Setting tab appears. See Figure 141.

4 Select the device from the Published **IP Address** drop-down list.

Figure 141 Global settings



Network Interfaces

Published IP Address

Published IP address: LAN1

Net Link Manager

Enable net link manager ☑

Primary interface: Permanent

Next hop on primary interface: 192.168.249.1

Backup interface: None

Up poll interval (s): 5

Down poll interval (s): 5

Switch-over delay (s): 30

Enable fast switch-over □

Modem

Modem region: US

Enable modem ☑

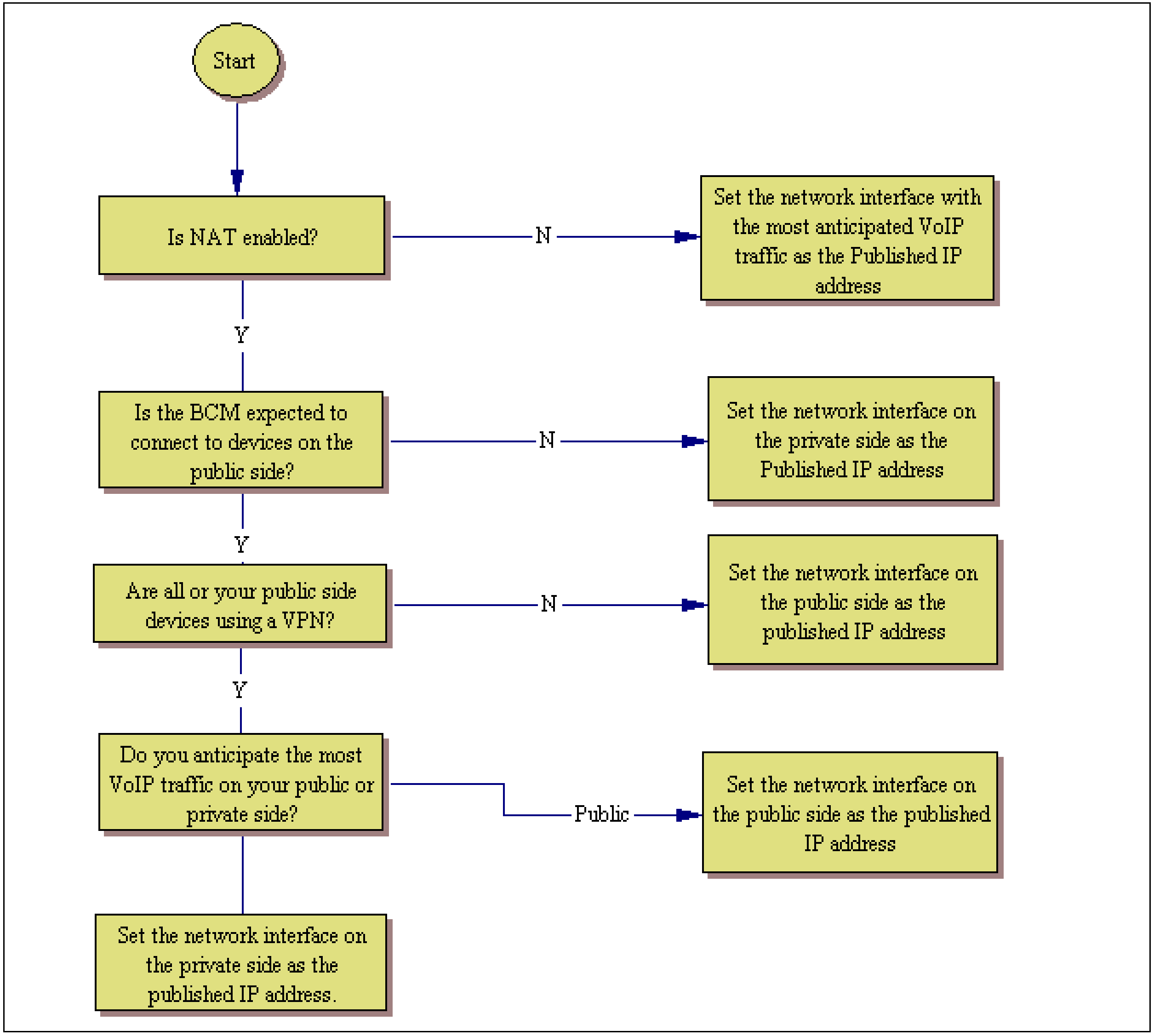
Table 117 Published IP Address options

Option	Description
LAN1	Choose the LAN number that corresponds with the LAN card you are using for this network.
LAN 2	
WAN1	Choose the WAN number that corresponds with the WAN card you are using for this network.
WAN2	

Determining the published IP address

Use the flowchart in the following figure to determine which card should be set as the published IP address.

Figure 142 Selecting the Published IP address


Start
Is NAT enabled?
N
Set the network interface with the most anticipated VoIP traffic as the Published IP address
Y
Is the BCM expected to connect to devices on the public side?
N
Set the network interface on the private side as the Published IP address
Y
Are all or your public side devices using a VPN?
N
Set the network interface on the public side as the published IP address
Y
Do you anticipate the most VoIP traffic on your public or private side?
Public
Set the network interface on the public side as the published IP address
Set the network interface on the private side as the published IP address.

The flowchart shown above makes reference to public and private IP addresses. The public and private IP addresses are concepts relating to Network Address Translation (NAT). The decision also depends on whether a Virtual Private Network (VPN) is enabled. For information about NAT and VPN, refer to *"Configuring NAT (Network Address Translation)" on page 607*.

If you use IP telephones on the network, they must be set to have the IP address of the network card they are connected to for their Default Gateway, and the Published IP address as the S1 IP address.

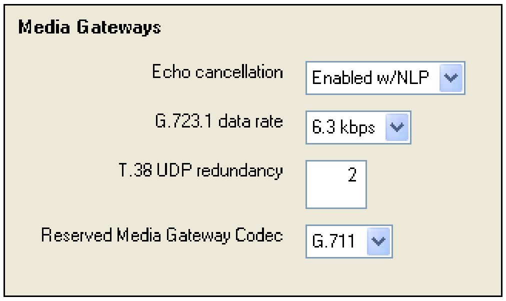
Media gateway parameters for IP service

To set up the media gateway resources that you require for optimum IP telephony and VoIP trunk service, you need to define some basic gateway parameters. These parameters are set in the **Media Gateways** panel.

Follow these steps to configure the media gateway:

1. Click **Configuration > Resources > Media Gateways.** See Figure 143.
2. Configure the settings according to Table 118.

Figure 143 Media Gateways Panel



Media Gateways

Echo cancellation: Enabled w/NLP

G.723.1 data rate: 6.3 kbps

T.38 UDP redundancy: 2

Reserved Media Gateway Codec: G.711

Table 118 IP terminals general record fields (Sheet 1 of 2)

Field	Value	Description
Echo cancellation	Enabled w/NLP Enabled Disabled	Enable or disable echo cancellation for your system. Default: Enabled w/NLP (check with your internet system administrator before changing this) Echo Cancellation selects what type of echo cancellation is used on calls that go through a Media Gateway. NLP refers to Non-Linear Processing.
G.723.1 data rate	6.3 kbps 5.3 kbps	Specify the data rate at which to process calls. This setting is not negotiated; it is manually administered and the far end must support the rate. The rate selection is based on quantity of calls, or quality of voice. • 5.3 kbps is a smaller bandwidth therefore more calls can be supported but the voice quality is lower. • 6.3 kbps is a larger bandwidth so fewer calls are supported but the voice quality is better. Default: 6.3 kbps
T.38 UDP Redundancy	0-3	This setting defines the number of times the system will transmit a UDP packet over the network. This setting acts as an error control mechanism for unreliable networks by providing the same information more than once, with the intention that at least one of the copies will transmit correctly. **WARNING:** Each redundancy requires the same amount of bandwidth as the original message. This means that a redundancy of 3 requires four times the bandwidth of a single transmission.

Table 118 IP terminals general record fields (Sheet 2 of 2)

Field	Value	Description
Reserved Media Gateway Codec	G.711 G.729 G.723	Choose the preferred codec that you are using with your IP network. Reserved Media Gateway Codec should be set to whatever is the most-commonly used codec for Media Gateways. It determines the amount of codec resources reserved for each Media Gateway. Reserving resources speeds up establishment of connections. For example, if most calls through a Media Gateway use the G.711 codec, set this to G.711. If most calls use G.729, set this to G.729. Note that the higher the setting (G.723 > G.729 > G.711) the more resources are set aside for Media Gateways. This may result in calls failing to go through because of lack of available resources.
For a more detailed descriptions of the media gateway or other information about the media services card (MSC) settings for the BCM, refer to the "Configuring application resources" on page 81.		

VoIP trunks

Complete this checklist if you are configuring VoIP trunks.

Table 119 VoIP trunk provisioning

Prerequisites	Yes	No
7.a Have you confirmed the remote gateway or Gatekeeper settings and access codes required? (H.323 and SIP trunks).		
7.b Have you determined the preferred codecs and payload sizes required for each type of trunk and destination?		
7.c Have you determined how you are going to split your VoIP resources between H.323 and SIP trunks.		
7.d Have you set up line parameters, determined line pools for H.323 and/or SIP trunks, and set up routing and destination codes? Have you determined which system telephones will have access to these routes?		
7.e If you have not already assigned target lines, have you defined how you are going to distribute them on your system?		
7.f Have you decided if you are going to employ the fallback feature? If yes, ensure that your routing and scheduling are set up. Ensure that QoS is activated. **Network note:** If your BCM is part of a private network, have the other BCMs in the network been upgraded to BCM 3.5 or newer software or had QoS patch 3.0.0.25 (or later) applied? If there is a Meridian 1 on the network, is it running IPT 3.0 or newer? If either of these conditions are not met, your H.323 trunks will not work correctly.		
Refer to “VoIP trunk gateways” on page 395, “Local gateway programming” on page 397, and “Optional VoIP trunk configurations” on page 404 for detailed configurations.		

IP telephone records

Complete this check list if you are installing Nortel IP telephones or WLAN wireless IP telephones.

Table 120 IP telephone provisioning (Sheet 1 of 2)

Prerequisites	Yes	No
8.a Are IP connections and IP addresses available for all IP telephones?		
8.b If DHCP is not being used, has all telephone configuration been documented and made available for telephone installers? **Hint:** Use the Programming Record form.		
8.c If DHCP is not being used, or if you want to enter the port manually, has the VLAN port number been supplied, if one is being used on the switch?		

Table 120 IP telephone provisioning (Sheet 2 of 2)

Prerequisites	Yes	No
8.d Have you determined the default codecs (and payload sizes), and default jitter buffers required by the IP network that supports the telephone?		
8.e Have telephone power and connectors been provisioned?		
8.f Do computers that will be using the Nortel 2050 Software Phone meet the minimum system requirements, including headset?		
8.g Do you want the system to auto-assign DNs? If no, complete 8.h. **Note:** If your company is using the Contact Center application on the BCM, Nortel recommends that you manually assign DNs to avoid conflicts with Contact Center DN assignments.		
8.h Have DN records been programmed for the corresponding IP clients? (Use when manually assigning DNs to the telephones.)		
8.i **WLAN wireless IP telephones**: Have you obtained the configuration program from the Documentation CD? Ensure that you have the correct IP addresses for the SVP server and for the TFTP server, if there is one. Decide if you want to allow the telephones to use DHCP to assign IP addresses, or enter static IP addresses.		
WLAN wireless handsets: Configuration and registration information is contained in separate documentation. Refer to the *WLAN IP Telephony Installation and Configuration Guide* (N0060634).		

Chapter 59
Configuring the LAN resources

BCM is equipped with an Ethernet/802.3 network interface card which supports the IEEE 802.3 Ethernet frame format. The Ethernet connection uses Carrier Sense Multiple Access with Collision Detection (CSMA/CD) to manage the access to the physical media.

Refer to the following topics for information about:

- "LAN resources" on page 487
- "Configuring LAN resources" on page 489

The BCM Ethernet interface card supports the following features:

- 100 BASE T with RJ-45 connector
- 10 / 100 Auto Sense
- full duplex
- multiple IP addresses

LAN resources

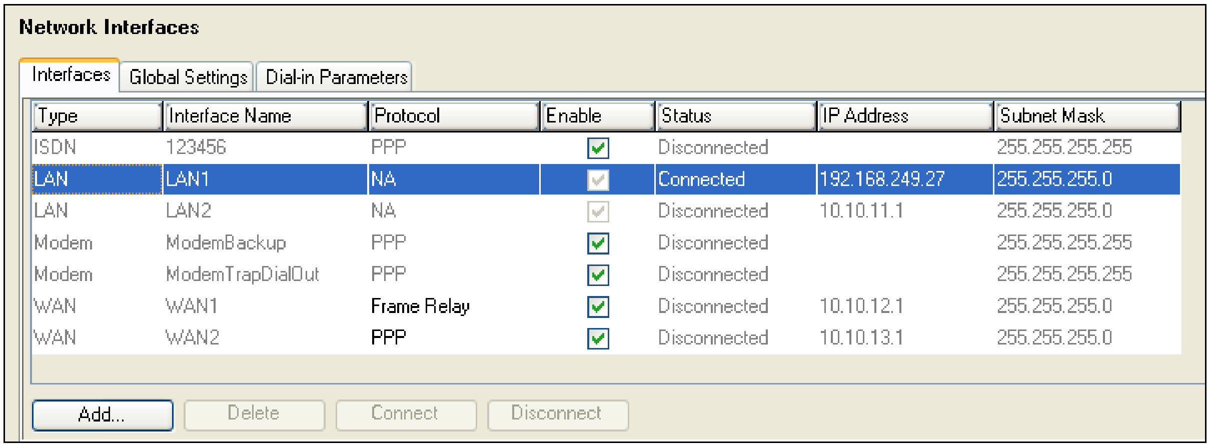
Element Manager shows all available LAN resources. If your BCM is equipped with two LAN interface cards, Element Manager displays all available LAN resources and names each one (LAN1, LAN2).

To view the available LAN resources

1. Click **Configuration > Resources > Network Interfaces**.
   The **Network Interfaces** panel appears.
2. Click **LAN1** or **LAN2**.
   The details panel appears.

Network Interfaces panel

Figure 144 Network Interfaces table



Type	Interface Name	Protocol	Enable	Status	IP Address	Subnet Mask
ISDN	123456	PPP	☑	Disconnected		255.255.255.255
LAN	LAN1	NA	☑	Connected	192.168.249.27	255.255.255.0
LAN	LAN2	NA	☑	Disconnected	10.10.11.1	255.255.255.0
Modem	ModemBackup	PPP	☑	Disconnected		255.255.255.255
Modem	ModemTrapDialOut	PPP	☑	Disconnected		255.255.255.255
WAN	WAN1	Frame Relay	☑	Disconnected	10.10.12.1	255.255.255.0
WAN	WAN2	PPP	☑	Disconnected	10.10.13.1	255.255.255.0

Table 6 Network Interfaces table fields

Attribute	Value	Description
Type	LAN WAN Modem ISDN	Displays the type of interface installed.
Interface Name	<read-only>	The name of the interface (LAN1 or LAN2).
Protocol	NA	The LAN protocol is not configurable.
Enable	<read-only>	Select to enable a dial-up interface. LAN and WAN check boxes are read-only.
Status	<read-only>	Shows the current status of the LAN connection. The possible states are: connected, disconnected
IP Address	<read-only>	Enter the IP address of the LAN interface. The IP address must be in the following format: 255.255.255.255.
Subnet Mask	<read-only>	Enter the subnet mask address of the LAN. The subnet mask IP address must be in the following format: 255.255.255.255.
Actions		
Add	<button>	Click to add a new network interface.
Delete	<button>	Select the network interface to delete from the interfaces table. A confirmation dialog appears to confirm the deletion.

Configuring LAN resources

Refer to the following information on how to set up the LAN card on your BCM:

- "To configure a LAN interface" on page 489
- "Configuring multiple IP addresses for the LAN interface" on page 493

To configure a LAN interface

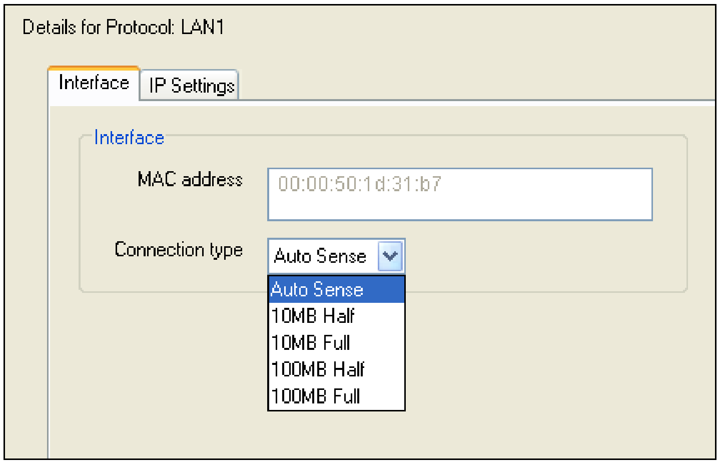
1 Click **Configuration > Resources > Network Interfaces**.
The Interfaces table appears.

2 Click the LAN interface you want to configure (for example, **LAN1**).
The LAN details panels appear.

3 Configure the LAN attributes. Refer to the information provided in Table 7 and Table 8.

LAN interfaces

Figure 145 LAN interface tab



Details for Protocol: LAN1

MAC address 00:00:50:1d:31:b7

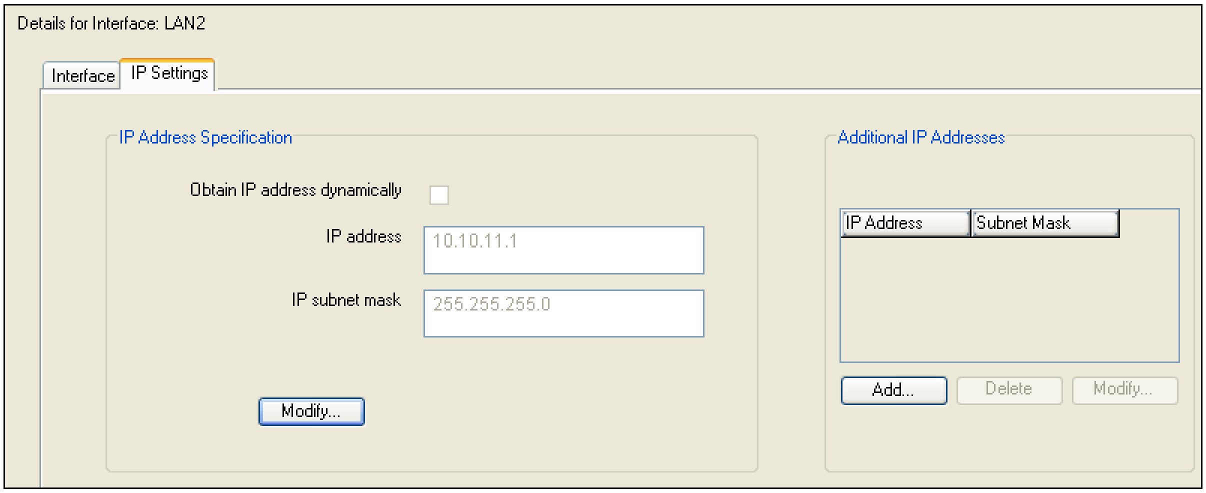
Connection type Auto Sense

Table 7 LAN Interfaces details

Attribute	Value	Description
Interface		
MAC address	<read-only>	Shows the physical address (MAC address) of the LAN interface.
Connection type	Auto Sense 10MB Half 10MB Full 100MB Half 100MB Full	Select a type of connection to the LAN interface. The following values are supported and are interpreted as follows: **Auto Sense**: The LAN interface uses the auto negotiation protocol to choose the maximum possible speed of the connection. Depending on the connected device, the LAN can choose 100 MB or 10MB and full-duplex or half-duplex. **10MB Half**: The speed is set to 10 Mbit/s and the mode is set to half-duplex. **10MB Full**: The speed is set to 10 Mbit/s and the mode is set to full-duplex. **100MB Half**: The speed is set to 100 Mbit/s and the mode is set to half-duplex. **100MB Full**: The speed is set to 100 Mbit/s and the mode is set to full-duplex. **Important**: If you have a 20XX IP telephone on your network, you must set the Connection Type to **Auto Sense**. **Note**: You may want to limit the incoming traffic to 10 Mbit/s if you notice that the busy traffic from the connected LAN is degrading the quality of voice calls carried through VoIP over the WAN. Though the LAN traffic gets lower priority in BCM, high incoming LAN traffic to the Business Communications Manager base unit can result in service interruptions in the system These interruptions may degrade the quality of voice calls carried as VoIP. Default: Auto Sense

IP Settings

Figure 146 LAN IP Settings



Details for Interface: LAN2

Obtain IP address dynamically □

IP address 10.10.11.1

IP subnet mask 255.255.255.0

Table 8 IP Settings

Attribute	Value	Description
IP Address Specification		
Obtain IP address dynamically	<check box>	If selected, enables DHCP. Default: Disabled
IP address	<IP address>	Enter the IP address of the LAN interface in the following format: 255.255.255.255. If you do not know your LAN interface IP address, contact your network administrator or your Internet service provider. **Web Cache:** If you have enabled Web Cache, you must update the Web Cache page when you change the IP address of the interface. Refer to "Web Cache page update required" on page 686. **IP Phones (Phase 2):** If you change the IP address of the BCM, any Phase 2 IP Phones using DHCP will lose their connection to the BCM. You must reboot the Phase 2 IP Phones to reconnect to the BCM.
IP subnet mask	<IP address>	Enter the subnet mask of the LAN interface in the following format: 255.255.255.255. If you do not know your subnet mask address, contact your network administrator or your Internet service provider.
Actions		
Modify	<button>	Click to modify the IP settings.

Note: Consult your network administrator for the appropriate configuration information before changing the settings.

Note: When you change these parameters, you must reboot the BCM system.

➡ **Note:** Setting the LAN connection speed to 100 Mbit/s does not reduce performance. However, the CPU is more efficient if you limit your incoming traffic to 10 Mbit/s. To increase your CPU performance, set the connected external LAN hub or switch to **10 Mbit/s** or to **Auto Sense**.

Configuring the LAN as a DHCP Client

The LAN on the BCM can be configured as a DHCP Client. The following connection types are supported:

- Auto Sense
- 10MB Half
- 10MB Full
- 100MB Half
- 100MB Full

Multiple addresses per LAN interface will be supported.

To configure the LAN as a DHCP client

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the LAN interface to configure.
   The details panel for that interface appears.
3. Click the **IP Settings** tab.
4. Click Modify in the **IP Address Specification** details panel.
   The **Modify IP Settings** dialog box appears
5. Select the **Obtain IP address dynamically** check box.
6. Click **OK**.

Configuring multiple IP addresses for the LAN interface

➡ **Note:** Refer to “Additional IP addresses are not blocked” on page 686 for important information regarding additional IP addresses for the LAN interface.

Adding an additional IP address

To add an additional IP address

1 Click **Configuration > Resources > Network Interfaces**.

2 Click the LAN resource you want to modify.
The LAN details panel appears.

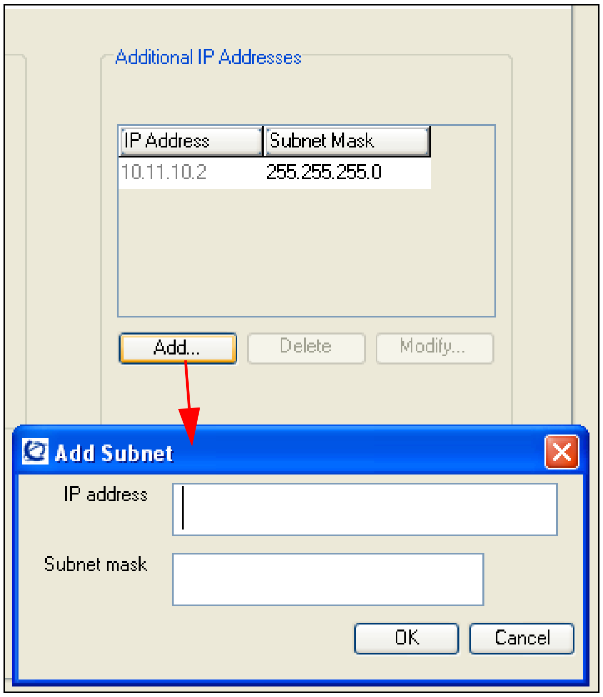
3 Click the **IP Settings** tab.
The Additional IP Address details panel appears.

4 Click **Add**.
The **Add Subnet** dialog box appears.

5 Configure the Additional IP Address parameters. Refer to the information in Table 9.

6 Click **OK.**

Figure 147 Additional IP Addresses



IP Address	Subnet Mask
10.11.10.2	255.255.255.0

Table 9 Add Additional IP Addresses parameters

Attribute	Value	Description
IP address	<IP address>	Enter the Additional IP address of the LAN interface.
Subnet mask	<IP address>	Enter the subnet mask of the LAN interface. If you do not know your subnet mask address, contact your system administrator or your Internet service provider.

Additional IP Addresses

Note: Each LAN interface can support up to five IP addresses.

To modify an additional IP Address

1 Click **Configuration > Resources > Network Interfaces**.

2 Click the LAN resource you want to modify.
The LAN Summary panel appears.

3 Click the **IP Settings** tab.
The **Additional IP Addresses** details panel appears.

4 Click the Additional IP Address you want to modify.

5 Click **Modify**.
The **Modify Subnet** dialog box appears.

6 Modify the IP Address or the Subnet Mask.

7 Click **Save**.

To delete an additional IP address

1 Click **Configuration > Resources > Network Interfaces**.

2 Click the LAN resource you want to modify.
The LAN Summary panel appears.

3 Click the **IP Settings** tab.
The **Additional IP Addresses** details panel appears.

4 Select the IP Address to delete.

5 Click **Delete**.
A confirmation dialog box appears.

6 Click **Yes**.

Chapter 60
Configuring the WAN resources

A WAN (wide area network) is a geographically dispersed data communication network. The term WAN distinguishes a broader data communication structure from a local area network (LAN).

Refer to the following topics:

- Permanent WAN connection
- "Viewing WAN resources" on page 499
- "Setting WAN T1 Parameters" on page 502
- "PVC Configuration" on page 507

A WAN can be privately owned or rented, but is usually part of public (shared user) networks.

BCM can be equipped with a WAN interface card with two serial synchronous ports (Europe), or a WAN interface card with one T1 port (with integrated CSU) and one serial synchronous port (North America). Both ports on the WAN interface card (WAN1 and WAN2) can be active at the same time. The serial synchronous port supports the following:

- North America: V.35
- Europe: V.35 (Upper Sync Port) and X.21 (Lower Sync Port)
- maximum line speed: 8 Mbit/sec.

BCM provides primary and backup WAN links through dial-up connections using a V.90 or V.92 modem or ISDN BRI/PRI. For information on the modem or ISDN connections, see "Configuring the Dial-up resources" on page 531. Net Link Manager provides continuous WAN connection status monitoring. For information about Net Link Manager, see "Configuring Net Link Manager" on page 561.

Permanent WAN connection

The permanent WAN connection is normally a dedicated network adapter. The permanent link supports Frame Relay or Point-to-Point protocol (PPP) at the link layer. The link protocol you use depends on the existing network or on the service you buy from your Internet service provider. The two ports provided by the WAN interface card can be independently configured to run Frame Relay or PPP.

Frame Relay

BCM supports frame relay in group mode and direct mode. In group mode, for each physical port (serial sync or T1 port), there is one IP address for all PVC*s* (permanent virtual circuits).

The available Data Link Connection Identifier numbers are 16-1007. The maximum number of PVCs supported is 50.

Point-to-Point Protocol (PPP)

Point-to-Point Protocol (PPP) is a full-duplex transmission protocol for communication between two computers using a serial interface. A typical PPP connection is a personal computer connected by telephone line to a server. For example, your Internet service provider (ISP) provides you with a PPP connection so that the ISP server can respond to your requests, pass them on to the Internet, and return your requested Internet responses to you.

- Multi-link Point-to-Point Protocol (MLPPP)

MLPPP is used to connect multiple B-channel*s* together when using PRI or BRI ISDN as the WAN interface. This allows BCM to connect B-channels independently of each other so that the ISDN connection can be used for both voice and data.

WAN data compression

BCM provides a WAN Data Compression feature. You can use data compression on a permanent WAN connection and on a backup WAN connection. You can enable or disable WAN Data Compression from the "Setting WAN Frame Relay Parameters" on page 505 panel or from the "WAN PPP Parameters" on page 510 panel, depending on your system configuration.

On a permanent WAN connection, BCM supports the following data compression protocols:

- Frame Relay Forum standard FRF.9 data compression protocol with STAC compression algorithm
- PPP Stac LZS Compression Protocol (RFC 1974) with STAC compression algorithm.

Note: Fragmentation and header compression cannot be enabled at the same time.

On dial-up WAN connections, BCM supports the following data compression protocol:

- Microsoft Point-to-Point Compression (MPPC), RFC 2118
- Header compression - This feature provides the option to compress the 40 byte IP/UDP/RTP header into 2 or 4 bytes on a link-by-link basis. By applying this compression over a slower link, it increases the bandwidth utilization significantly. As a result, more concurrent VoIP trunks can be put into the same link at the same time, and the voice quality is improved because of the reduced transmission delay. The IP/UDP/RTP Header Compression is implemented based on RFC 2508. Protocol negotiation of IP/UDP/RTP header compression is based on RFC 2509 in PPP and FRF .20 in Frame Relay.
- packet fragmentation - Fragmentation is a required feature for all Nortel network products if the products are operated over network links with bandwidth of 1M bps or lower. The major advantage of doing link-layer fragmentation (such as Multilink PPP) over the IP layer fragmentation over a slow link is: while it reduces the jitter effect for voice packets, it also reduces the end-to-end delay for data packets as introduced by the IP layer fragmentation. The Multilink PPP implementation follows RFC 1990.

- Multilink PPP, by default, defines the usage of Short Sequence Numbered (12 bits) fragment. Additionally Long Sequence Numbered (24 bits) Fragment as defined in RFC 2686 is also supported over the Multilink PPP.
- The Frame Relay fragmentation implementation follows FRF.12 Specification.
- suitable for VoIP

Viewing WAN resources

To view available WAN resources

1 Click **Configuration** > **Resources** > **Network Interfaces** and click the **WAN** heading. The details panel appears.

The Interfaces panel shows the Type, Interface Name, Protocol, Status, IP address, and Subnet Mask of all of the WAN interfaces on the BCM.

Note: If you disconnect the WAN cable from the WAN card, the Status does not update immediately. It can take more than two minutes before the Status updates to show the new Status of the WAN card.

Configuring the WAN interfaces

Refer to the following information to configure the WAN interfaces on the BCM system.

This includes the following:

- "Configuring WAN summary parameters" on page 499
- "Setting WAN T1 Parameters" on page 502
- "Setting WAN Sync Parameters" on page 504
- "Setting WAN Frame Relay Parameters" on page 505
- "PVC Configuration" on page 507
- "WAN PPP Parameters" on page 510
- "Multilink PPP Parameters" on page 513
- "Configuring multiple IP addresses for a WAN interface" on page 515
- "Configuring the DLCI to IP Mapping" on page 517

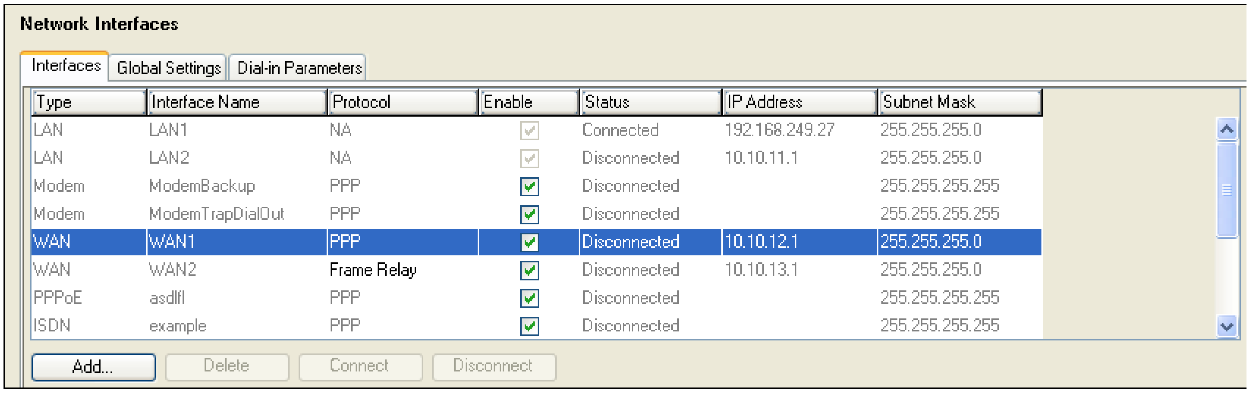
Configuring WAN summary parameters

1 Click **Configuration** > **Resources** > **Network Interfaces**.

2 Click the **WAN1** or **WAN2** heading. See Figure 148.

3 Configure the WAN settings according to the information in Table 10.

Figure 148 WAN Summary parameters



Type	Interface Name	Protocol	Enable	Status	IP Address	Subnet Mask
LAN	LAN1	NA	☑	Connected	192.168.249.27	255.255.255.0
LAN	LAN2	NA	☑	Disconnected	10.10.11.1	255.255.255.0
Modem	ModemBackup	PPP	☑	Disconnected		255.255.255.255
Modem	ModemTrapDialOut	PPP	☑	Disconnected		255.255.255.255
WAN	WAN1	PPP	☑	Disconnected	10.10.12.1	255.255.255.0
WAN	WAN2	Frame Relay	☑	Disconnected	10.10.13.1	255.255.255.0
PPPoE	asdlfl	PPP	☑	Disconnected		255.255.255.255
ISDN	example	PPP	☑	Disconnected		255.255.255.255

Table 10 WAN summary parameters

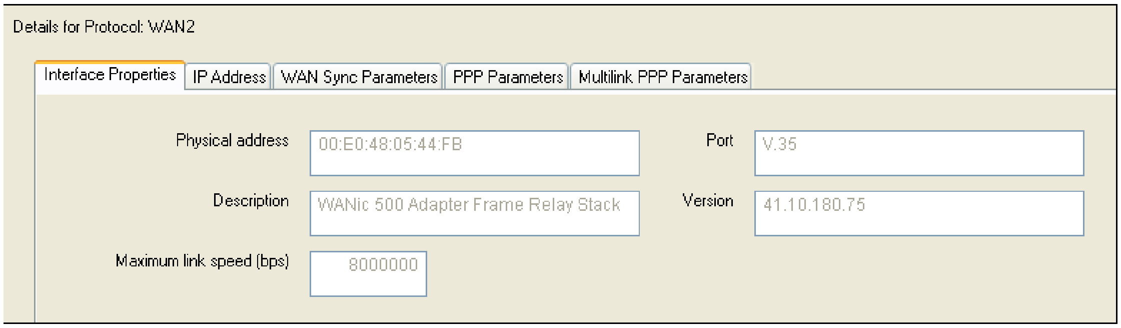
Attribute	Value	Description
Type	<read-only>	Identifies the types of devices connected.
Interface Name	<read-only>	The name of the interface (WAN1 or WAN2).
Protocol	Frame Relay PPP	Select a WAN link protocol. The options are **Frame Relay** or **PPP** protocol. If you change the link protocol, the configuration panel changes to include fields corresponding to the link protocol you choose. To ensure proper operation, always refresh the page by clicking **View** and then **Refresh**. The link protocol you choose depends on the existing network or the service you buy from your Internet services provider. Default: Frame Relay
Enable	<check box>	Shows the current resource status of the WAN interface. The possible states are: **Selected:** The WAN card is operational. **Cleared**: The WAN card is not operational.
Status	<read-only>	Displays the status of the device.
IP Address	<read-only>	The IP address of the WAN interface. There are always two WAN interfaces, WAN1 and WAN2. Each WAN interface can be configured with its own IP Address and Subnet Mask value in the IP Address settings tab. You can obtain this information from your system administrator or your Internet service provider. **Note:** If you have enabled Web Cache, you must update the Web Cache page when you change the IP address of the interface. Refer to "Web Cache page update required" on page 686.
Subnet Mask	<read-only>	The subnet mask address of the WAN interface. The subnet mask IP address must be in the following format: 255.255.255.255. You can obtain this information from your system administrator or your Internet service provider.

Note: Element Manager refreshes the link protocol panel according to the chosen protocol. Your choice of protocol depends on the existing network or the service you buy from your Internet service provider. Frame relay is the default link protocol. If you change the link protocol the following message appears “The system needs to be rebooted for the changes to take effect. Please reboot the system” Click **OK**.

Caution: Reboot the system
You must remember to reboot your system for the changes you made to the link protocol to take effect. You can continue Resources configuration and reboot the system at a convenient time.

WAN Interface Properties

Figure 149 WAN Interface Properties



Details for Protocol: WAN2

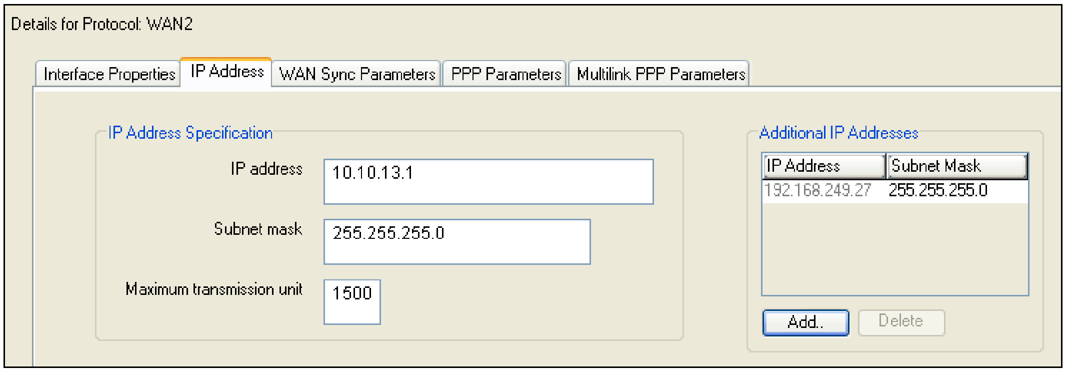
Physical address: 00:E0:48:05:44:FB
Port: V.35
Description: WANic 500 Adapter Frame Relay Stack
Version: 41.10.180.75
Maximum link speed (bps): 8000000

Table 11 WAN Interface Properties fields

Attribute	Value	Description
Physical address	<read-only>	Physical address of the WAN interface.
Description	<read-only>	Description of the Network interface card that supports the WAN connection.
Maximum link speed (bps)	<read-only>	Operation speed of the WAN interface.
Port	<read-only>	Port type of the WAN interface.
Version	<read-only>	Version of the WAN interface.

WAN IP Address

Figure 150 WAN IP Address



Details for Protocol: WAN2

IP Address Specification

IP address 10.10.13.1

Subnet mask 255.255.255.0

Maximum transmission unit 1500

Additional IP Addresses

IP Address	Subnet Mask
192.168.249.27	255.255.255.0

Table 12 WAN IP Address fields

Attribute	**Value**	**Description**
IP Address Specification		
IP address	<IP address>	Specify the IP address of the WAN.
Subnet mask	<IP address>	Specify the subnet mask.
Maximum Transmission Unit	<256-4096>	The Maximum Transmission Unit (MTU) is the largest size of IP datagram which may be transferred using this data link connection. Default: 1500.
Additional IP Addresses		
Actions		
Add	<button>	Click to add an additional IP address.
Delete	<button>	Select an IP address to delete and click.

Setting WAN T1 Parameters

The WAN T1 Parameters panel is displayed when configuring a T1 port (North America only). BCM supports T1 and fractional T1.

To set WAN T1 parameters

Note: The WAN T1 Parameters panel is only available on the WAN1 interface.

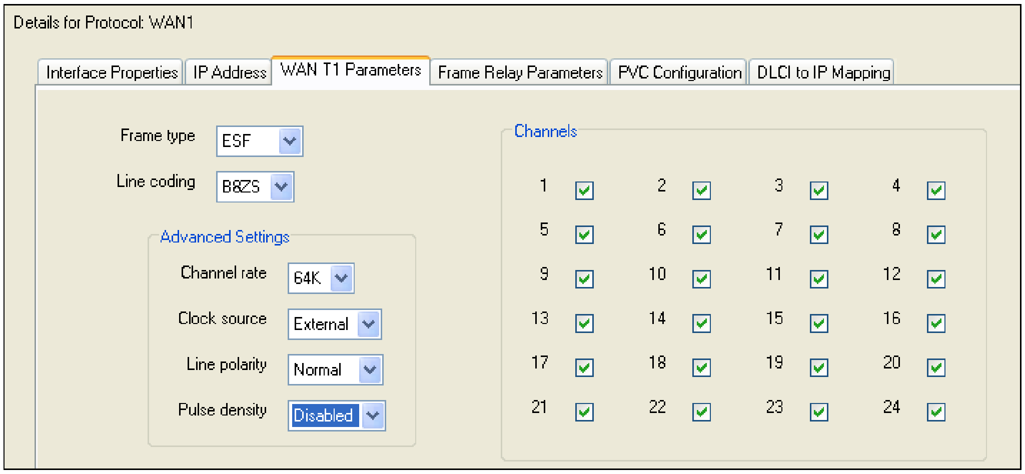
1 Click **Configuration > Resources > Network Interfaces**.

2 Click the **WAN1** heading.
The WAN details panel appears.

3 Click the **WAN T1 Parameters** tab.
The WAN T1 Parameters details panel appears. See Figure 151.

4 Configure the WAN T1Parameters. Refer to the information in Table 13.

Figure 151 WAN T1 parameters



Details for Protocol: WAN1

Frame type: ESF
Line coding: B8ZS

Advanced Settings
Channel rate: 64K
Clock source: External
Line polarity: Normal
Pulse density: Disabled

Channels

1 ☑	2 ☑	3 ☑	4 ☑
5 ☑	6 ☑	7 ☑	8 ☑
9 ☑	10 ☑	11 ☑	12 ☑
13 ☑	14 ☑	15 ☑	16 ☑
17 ☑	18 ☑	19 ☑	20 ☑
21 ☑	22 ☑	23 ☑	24 ☑

Table 13 WAN T1 parameters

Attribute	**Value**	**Description**
Frame type	ESF SF(D4)	Set the type of framing the T1 line supports. Use the setting your T1 service provider recommends. Default: ESF
Line coding	B8ZS AMI	Set the type of encoding used in the T1 line. Use the setting your T1 service provider recommends. Default: B8ZS
Advanced Settings		
Channel rate	56K 64K	Set the data transmission rate for each of the DS0 channels in the T1 line. Default: 64 K
Clock source	External Internal	Set an internal or external T1 clock source. For a tandem connection, if one end has the clock source as Internal, then the other end should have the clock set as external, as the farther end will supply the clock. Default: External
Line polarity	Normal Inverted	Set Normal or Inverted line polarity in the T1 line. Select Inverted only if Line Coding is set to AMI. Default: Normal
Pulse density	Enabled Disabled	Control whether the DSU/CSU maintains the minimum level of 1s on the line for AMI encoding. Default: Disabled
Channels	<check box>	Create a list of T1 channels used when using fractional T1. Your T1 service provider can give you this information. Default: All

➡ **Note:** Always use the same frame type and line coding method as your service provider.

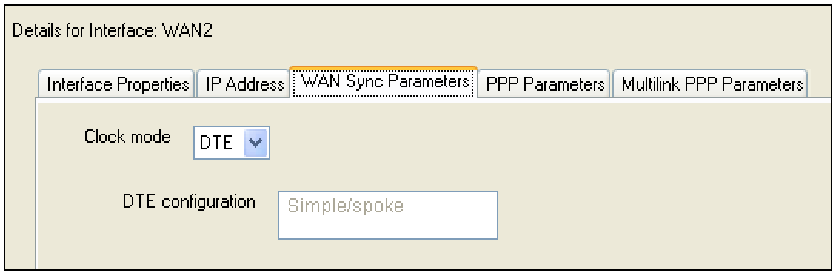
Setting WAN Sync Parameters

The WAN Sync Parameters panel is available for configuring the V.35 and the X.21 of the WAN card.

To set WAN Sync Parameters

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN2** heading.
   The WAN details panel appears.
3. Click the **WAN Sync Parameters** tab.
   The WAN Sync Parameters panel appears. See Figure 152.
4. Configure the WAN Sync Parameters. Refer to the information in Table 14.

Figure 152 WAN Sync Parameters



Details for Interface: WAN2

Clock mode DTE

DTE configuration Simple/spoke

Table 14 WAN Sync Parameters

Attribute	Value	Description
Clock mode	DTE DCE	Choose the clock mode.
DTE configuration	Simple/spoke	Read-only field.

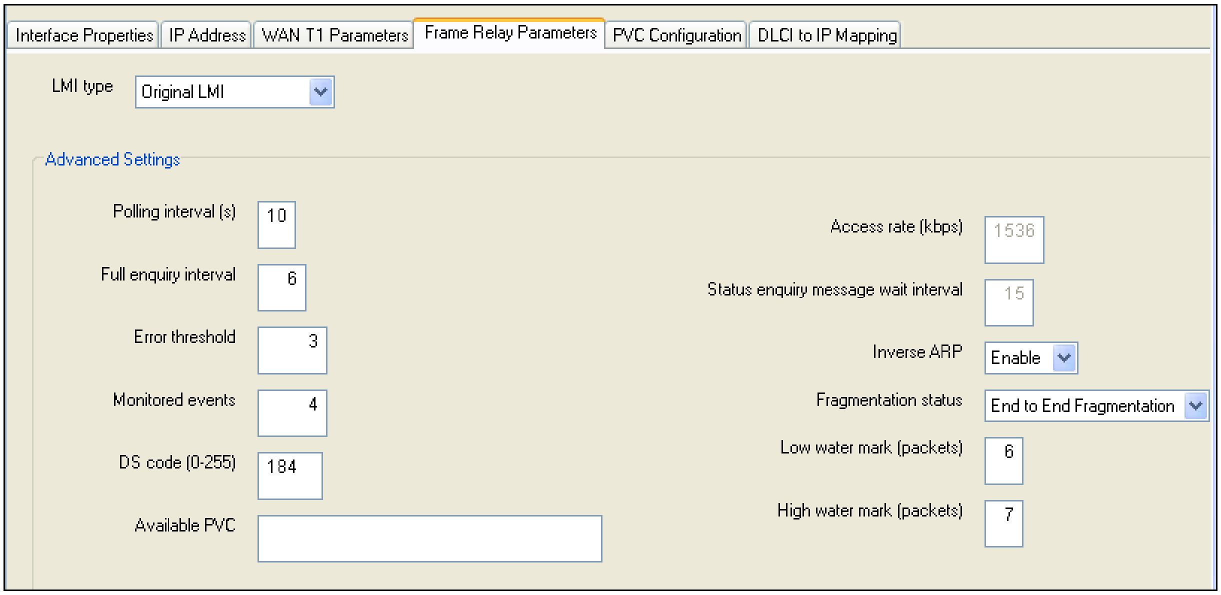
Setting WAN Frame Relay Parameters

To set WAN Frame Relay Parameters

If you chose frame relay as your link protocol, set the WAN Frame Relay Parameters.

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN details panel appears.
3. Click the WAN **Frame Relay Parameters** tab.
   The WAN **Frame Relay Parameters** details panel appears. See Figure 153.
4. Configure the WAN Frame Relay Parameters. Refer to the information in Table 15.

Figure 153 Frame Relay Parameters



LMI type: Original LMI

Advanced Settings

Polling interval (s): 10

Full enquiry interval: 6

Error threshold: 3

Monitored events: 4

DS code (0-255): 184

Available PVC:

Access rate (kbps): 1536

Status enquiry message wait interval: 15

Inverse ARP: Enable

Fragmentation status: End to End Fragmentation

Low water mark (packets): 6

High water mark (packets): 7

Table 15 WAN frame relay parameters (Sheet 1 of 2)

Attribute	Value	Description
LMI type	Original LMI ANSI T1.617 Annex D ITU-T Q.933 Annex A	Select the type of local management protocol used on this link. The link management type must be the same as the one used by the frame relay service provider. Note: The most commonly used setting for this parameter is ANSI T1.617 Annex D. Default: Original LMI
Advanced Settings		
Polling interval (s)	<5-30>	Enter an interval to wait between LMI status inquiry messages. The polling interval must be the same as the one used by the frame relay service provider's switch. Default: 10 seconds
Full enquiry interval	<1-255 >	Enter the maximum number of LMI Status Enquiry messages sent before sending a Full Status Enquiry request. This value must match the corresponding value set in the frame relay service provider's switch. Default: 6 seconds
Error threshold	<0-65000>	Enter the maximum number of eroded events (for example non-receipt of Status messages or receipt of Status messages with invalid sequence numbers) permitted by LMI, before dropping the connection. The error threshold is also the number of successful LMI Status messages that must be received before making the connection operational. If you have a backup WAN connection and Net Link Manager configured, the backup connection is started and traffic is routed to the backup when this link is dropped. Also, the backup WAN connection is dropped and traffic is routed to this link when the link is operational. For information about Net Link Manager, refer to "Configuring Net Link Manager" on page 561. Default: 3 failures

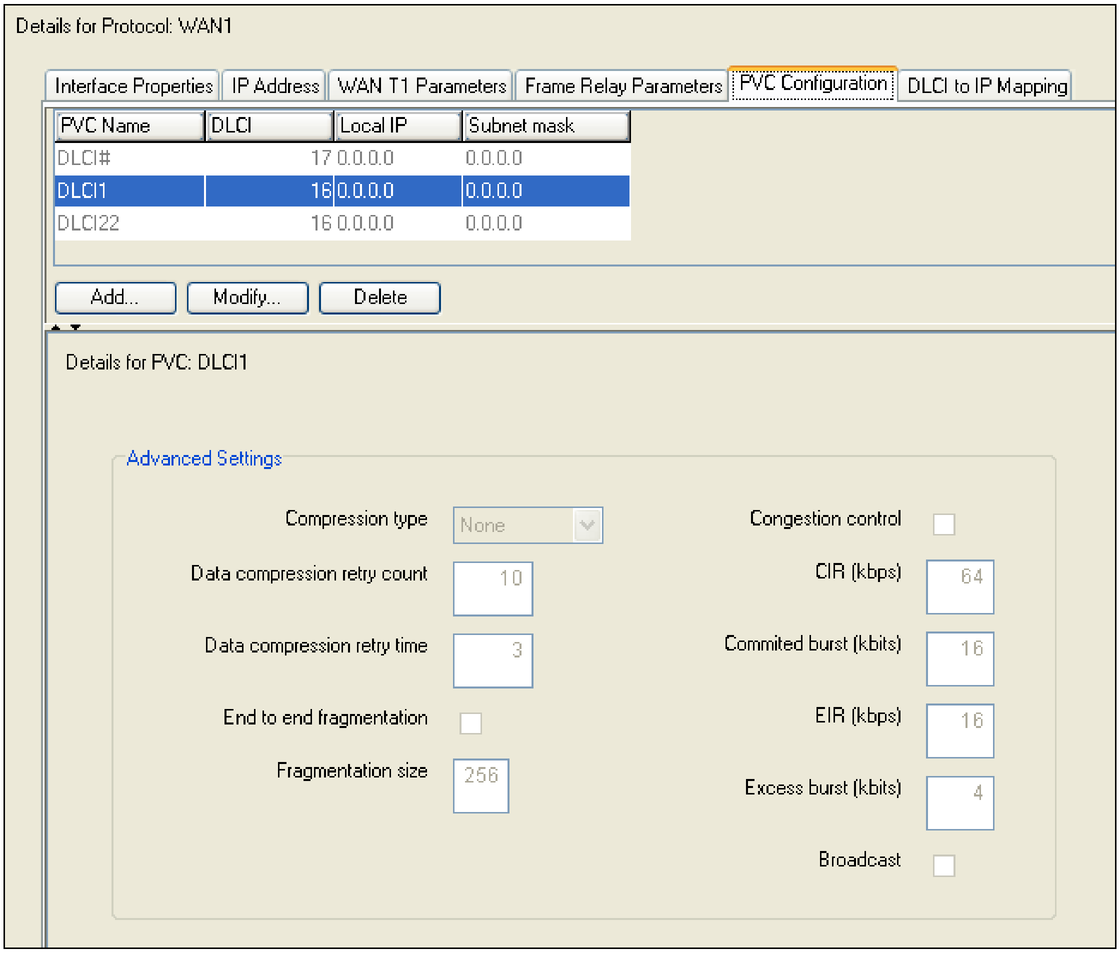
Table 15 WAN frame relay parameters (Sheet 2 of 2)

Attribute	**Value**	**Description**
Monitored events	<0-65000>	Enter the number of events sampled for making decisions about the error threshold. This value must be set to a higher number than the value set in the Error Threshold box. Default: 4
DS code (0-255)	<0-255>	Enter the Differentiated Services code recognized by the frame relay driver for traffic prioritization. When an IP packet is sent, the frame relay driver checks if the packet's ToS field (in the IP header) matches with the value defined in the DS Code. If the values match, then the packet is not treated as Discard Eligible (DE) packet during congestion. Default: 184
Available PVC	<read-only>	Lists the PVCs (Permanent Virtual Circuits) available for this WAN interface.
Access rate (kbps)	<read-only>	The available bandwidth on the interface running Frame Relay. The value is determined based on the number of channels selected in the WAN T1 Parameters tab and the channel rate value for T1 interfaces. If the interface type is Synchronous, the Access Rate value is determined based on the Clock Rate attribute for DCE interfaces or based on the Configured Linkspeed for DTE interfaces located under **Configuration > Data Services > QoS Queuing**.
Status enquiry message wait interval	<1-100>	Select the interval to wait for the arrival of the status enquiry message. Default: 15 seconds
Inverse ARP	Enable Disable	Disabled - the remote end IP address and DLCI mapping has to be provisioned. Enabled - the remote end IP address and DLCI mapping is discovered. Default: Enable
Fragmentation status	Disable End to End fragmentation	Controls the Frame Relay end-to-end fragmentation on the interface. **Note:** Enable the end-to-end fragmentation at PVC level to fragment the frames sent over the PVC. Default: End to End fragmentation
Low water mark (packets)	<1-14>	Lower threshold that controls the link layer queuing behavior. Default: 6
High water mark (packets)	<2-15>	Upper threshold that controls the link layer queuing behavior. Default: 7

PVC Configuration

If frame relay is your link protocol, you must configure PVC. If PPP is your link protocol, there are no PVC Congestion Control settings to configure.

Figure 154 PVC Configuration



Details for Protocol: WAN1

PVC Name	DLCI	Local IP	Subnet mask
DLCI#	17	0.0.0.0	0.0.0.0
DLCI1	16	0.0.0.0	0.0.0.0
DLCI22	16	0.0.0.0	0.0.0.0

Details for PVC: DLCI1

Advanced Settings

Compression type: None

Data compression retry count: 10

Data compression retry time: 3

End to end fragmentation: □

Fragmentation size: 256

Congestion control: □

CIR (kbps): 64

Commited burst (kbits): 16

EIR (kbps): 16

Excess burst (kbits): 4

Broadcast: □

Table 16 PVC Configuration fields (Sheet 1 of 2)

Attribute	Value	Description
DLCI	<16-1007>	Enter the data link connection identifier (DLCI) number of the PVC on which to perform congestion control. A DLCI must be configured for congestion control to be performed. BCM uses one-second intervals to measure this parameter. Default: 16
Advanced Settings		
Compression type	None RTP Header Data	Select RTP Header option to enable IP/UDP/RTP Header compression over this PVC. Select Data option to enable STAC Data Compression over this PVC. Default: None
Data compression retry count	<1-128>	Enter the number of times to re-initiate Mode 1-DCPCP before terminating the handshaking procedure and entering the disabled state. Default: 10

Table 16 PVC Configuration fields (Sheet 2 of 2)

Attribute	Value	Description
Data compression retry time	<1-15>	Enter the interval to wait before restart handshake procedure by the Mode 1-DCPCP in case the remote end fails to respond. Default: 3 seconds
End to end fragmentation	<check box>	Select to Enable End-to-End Fragmentation on this PVC. Default: Disabled
Fragmentation size	<200-800>	Maximum frame size in bytes. Any packets that have a frame size larger than this value will be fragmented. Default: 256
Congestion control	<check box>	Parameter used for enabling Congestion Control per PVC. Default: Disabled
CIR (kbps)	<0-1024>	Enter the committed information rate in kbits. The CIR is the data rate, the carrier commits to the user, that the router transmits at over a predetermined time interval when congestion is not present. Contact your service provider for the correct setting. BCM uses one-second intervals to measure this parameter. Default: 64
Committed burst (kbits)	<read-only>	Maximum amount of data the carrier guarantees to transport over the network, under normal conditions, during the Tc measurement interval. Calculate this value with the formula: BC=CIRxTc (Tc=1/4 in BCM implementation).
EIR (kbps)	<read-only>	Excess Information Rate is determined by the excess burst divided by Tc. EIR=Be/Tc.
Excess burst (kbits)	<0-1024>	Excess burst (Be) is the maximum amount of data by which the user may exceed the committed burst during a Tc measurement interval. The combined value of committed burst and excess burst must be less than, or equal to, the line speed. Default: 16
Broadcast	<check box>	When broadcast is enabled, broadcast packets are sent out over this PVC.

To add PVC congestion control

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **PVC Configuration** tab.
   The PVC Configuration panel appears.
4. Click **Add.**
5. The **Add PVC Configuration** dialog box appears. See Figure 154.
6. Configure the WAN PVC parameters. Refer to the information in Table 15.
7. Click **OK**.

To modify a PVC setting

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **PVC Configuration** tab.
   The PVC Configuration panel appears.
4. Click the entry you want to modify in the PVC table.
5. Click **Modify**.
   The Modify PVC Configuration dialog box appears.
6. Change the PVC Configuration parameters. Refer to the information in Table 16.
7. Click **OK**.

To delete a PVC congestion control setting

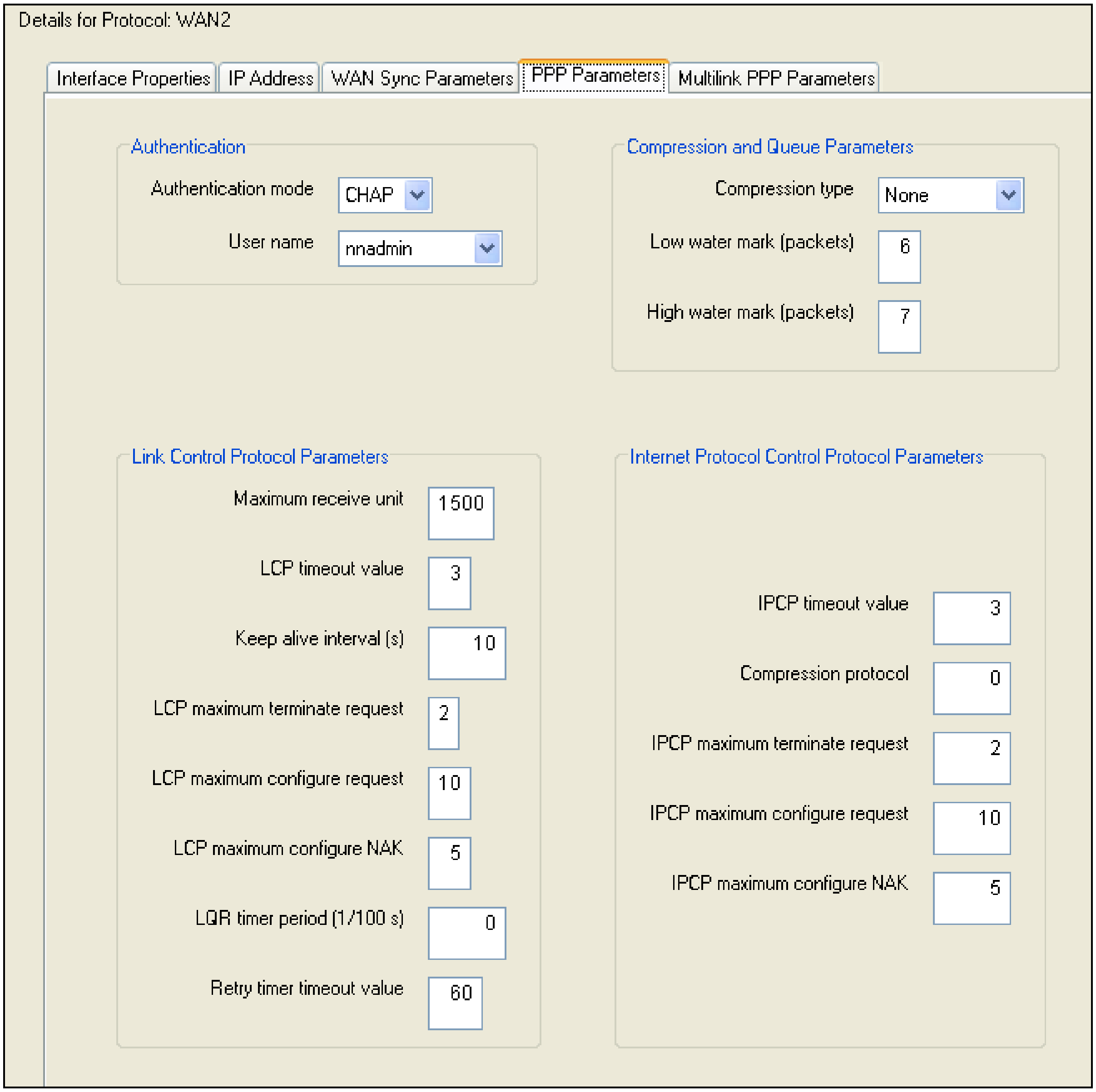
1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **PVC Configuration** tab.
   The PVC Configuration details appears.
4. Click the entry you want to delete in the PVC table.
5. Click **Delete**.
   A message prompts you to confirm the deletion.
6. Click **Yes**.

WAN PPP Parameters

If you chose PPP as your link protocol, set the WAN PPP Parameters panel.

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **PPP Parameters** tab.
   The WAN PPP Parameters panel appears. See Figure 155.
4. Configure the WAN PPP Parameters. Refer to the information in Table 17.

Figure 155 WAN PPP Parameters



Details for Protocol: WAN2

Authentication

Authentication mode: CHAP

User name: nnadmin

Compression and Queue Parameters

Compression type: None

Low water mark (packets): 6

High water mark (packets): 7

Link Control Protocol Parameters

Maximum receive unit: 1500

LCP timeout value: 3

Keep alive interval (s): 10

LCP maximum terminate request: 2

LCP maximum configure request: 10

LCP maximum configure NAK: 5

LQR timer period (1/100 s): 0

Retry timer timeout value: 60

Internet Protocol Control Protocol Parameters

IPCP timeout value: 3

Compression protocol: 0

IPCP maximum terminate request: 2

IPCP maximum configure request: 10

IPCP maximum configure NAK: 5

Table 17 WAN PPP parameters (Sheet 1 of 3)

Attribute	**Value**	**Description**
Authentication		
Authentication mode	PAP CHAP	Specify the Authentication mode a remote user can use. You can select **PAP** or **CHAP**. Select CHAP only to restrict the remote user to using CHAP authentication. Select PAP or CHAP to allow the remote user to use PAP or CHAP authentication. Default: CHAP

Table 17 WAN PPP parameters (Sheet 2 of 3)

Attribute	Value	Description
User name	nnadmin BTRemoteIsdn	Select the user name used for authentication by the far end of the PPP connection. Default: nnadmin
Compression and Queue Parameters		
Compression type	None RTP Header Data	Select RTP Header option to enable IP/UDP/RTP Header compression on this PPP Link Select Data option to enable STAC Data Compression on this PPP Link Default: None
Low water mark (packets)	<1-14>	Parameters that control the link layer queuing behavior for PPP when link layer fragmentation is enabled. Default: 6
High water mark (packets)	<2-15>	Parameters that control the link layer queuing behavior for PPP when link layer fragmentation is enabled. Default: 7
Link Control Protocol Parameters		
Maximum Receive Unit	<64-1614>	Maximum size of the received packets. Default: 1500
LCP timeout value	<1-60>	Time before retransmitting Configure-Request and Terminate-Request packets. Default: 3 seconds
Keep alive interval (sec)	<numeric>	This parameter determines the how long the BCM waits before determining that the WAN link is not functioning. When there is no regular traffic on the WAN link, the BCM sends an echo packet out on the link. If there is no response within the LCP Keep Alive Interval, the BCM sends another echo packet. If there is no response after 10 echo packets, the BCM determines that the link is down. If there is a response to any of the echo packets, the BCM determines that the WAN link is functioning. Enter the LCP Keep Alive Interval, in seconds, that the BCM waits between sending echo packets. Default: 10 seconds
LCP maximum terminate request	<1-5>	Maximum number of Terminate-Request packets sent. Default: 2
LCP maximum configure request	<1-15>	Maximum number of Configure-Request packets sent. Default:10
LCP maximum configure NAK	<1-10>	Maximum number of Configure-NAK packets sent. Default: 5
LQR timer period (1/100 s)	<0-65000>	Interval, in 1/100 second, to perform link quality monitoring. Default: 0

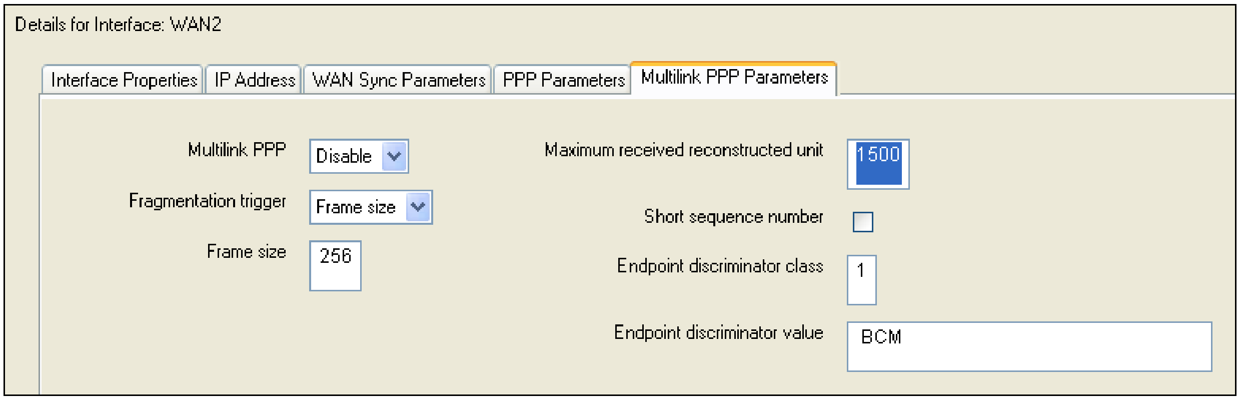
Table 17 WAN PPP parameters (Sheet 3 of 3)

Attribute	Value	Description
Retry timer timeout value	<30-180>	Time before retransmitting LCP Configure-Request packets. Default: 60 seconds
Internet Protocol Control Protocol Parameters		
IPCP timeout value	<0-65000>	Time before retransmitting Configure-Request and Terminate-Request packets. Default: 3 seconds
Compression protocol	<read-only>	This parameter is currently ignored and always FULL_HEADER_COMPRESSION (i.e. 0x0061) protocol is negotiated.
IPCP maximum terminate request	<0-65000>	Maximum number of Terminate-Request packets sent. Default: 2
IPCP maximum configure request	<0-65000>	Maximum number of Configure-Request packets sent. Default: 10
IPCP maximum configure NAK	<0-65000>	Maximum number of Configure-NAK packets sent. Default: 5

Multilink PPP Parameters

Multilink PPP layer fragmentation over a slow link reduces the jitter effect for voice packets, and reduces the end-to-end delay for data packets as introduced by the IP layer fragmentation.

Figure 156 Multilink PPP Parameters panel



Details for Interface: WAN2

Multilink PPP: Disable

Fragmentation trigger: Frame size

Frame size: 256

Maximum received reconstructed unit: 1500

Short sequence number: □

Endpoint discriminator class: 1

Endpoint discriminator value: BCM

Table 18 Multilink PPP Parameters panel details

Attribute	**Value**	**Description**
Multilink PPP	Disable Enable	Select to enable Multilink PPP on this interface The setting you choose for Fragmentation Status must be the same as the setting on the other end of the PPP link. Default: Disable
Fragmentation trigger	Frame size Delay Time	Select **Frame size** if you want BCM to use the size of the packet to decide if the packets need to be fragmented. Select **DelayTime** if you want BCM to use delay time to decide if the packets need to be fragmented. Default: Frame size
Frame size	<160-800>	Parameter is only available if you selected **Frame size** for the Fragmentation Trigger parameter. Enter the maximum frame size in bytes. Any packets that have a frame size larger than the size specified are fragmented. Default: 256
Maximum received reconstructed unit	<64-1614>	Maximum number of octets that can be in the information field of the reassembled packet. Default: 1500
Short sequence number	<check box>	Select if you want to receive fragments with short, 12-bit sequence numbers. Default: Disabled
Endpoint discriminator class	<0-5>	Multilink Class for the Endpoint Discriminator. Default: 1
Endpoint discriminator value		Multilink identifier address for the Endpoint Discriminator. Default: BCM

Configuring multiple IP addresses for a WAN interface

You can assign up to five IP addresses to a single WAN interface that is configured to use frame relay. Using this functionality, you can configure the BCM as the hub in a hub and spoke configuration. When BCM is the hub or central site, BCM can provide at least two IP address classes on the primary WAN interface. This allows the system to provide Direct Mode capability.

> **Note:** Refer to "Additional IP addresses are not blocked" on page 686 for important information regarding additional IP addresses for the WAN interface.

Examples of uses of multiple IP addresses

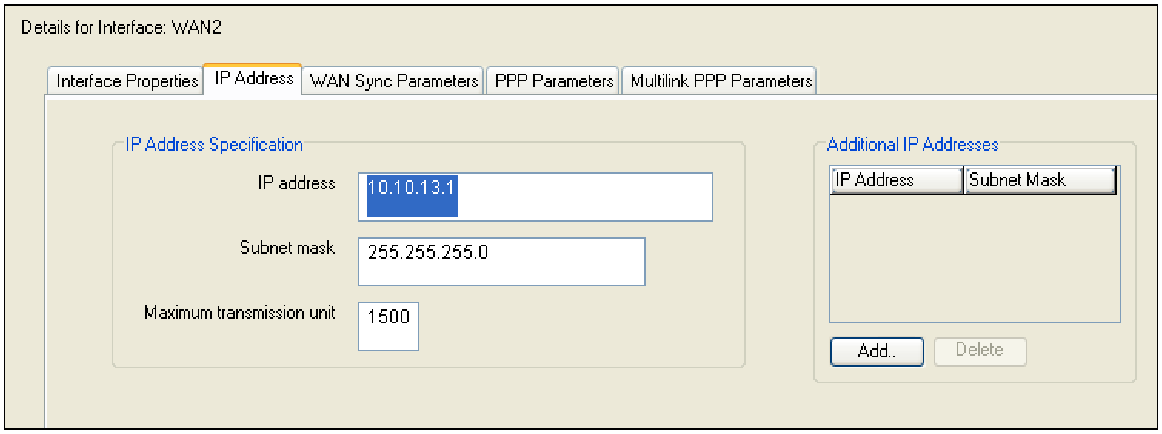
- You can use a single WAN physical link to connect to both an intranet and the internet using separate addressing schemes.
- A network service provider can create a separate IP address for management functions over the WAN interface.

In both of these examples, broadcast traffic destined for one IP address would not be transmitted on the links associated with the other IP address.

Restrictions when using multiple IP addresses

- Nortel does not recommend using more than two IP address classes.
- Multiple IP addresses supports RIP and OSPF.
- IPSec does not support the use of these multiple IP addresses for Branch Office Local Endpoint Addresses, Remote Endpoint Addresses or the Destination IP Address for IPSec VPN Clients.

Figure 157 Additional IP address parameters



Details for Interface: WAN2

IP address: 10.10.13.1

Subnet mask: 255.255.255.0

Maximum transmission unit: 1500

Table 19 Additional WAN IP addresses

Attribute	Value	Description
IP address	<IP address>	Enter the Additional IP address of the WAN interface in the following format: 255.255.255.255.
Subnet mask	<IP address>	Enter the subnet mask of the WAN interface in the following format: 255.255.255.255. If you do not know your subnet mask address, contact your system administrator or your Internet service provider.

To add an additional IP address

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **IP Address** tab.
   The Additional IP Address panel appears.
4. Click **Add**.
   The Add IP Address dialog box appears.
5. Configure the Additional IP Address parameters. Refer to the information in Table 19.
6. Click **OK**.

To modify an additional IP address

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the **WAN1** or **WAN2** heading.
   The WAN Summary panel appears.
3. Click the **IP Address** tab.
   The Additional IP Address details panel appears.

4 Click the IP Address you want to modify.

5 Click **Modify**.

6 Change the Additional IP Address parameters.

7 Click **OK**.

To delete an IP address

1 Click **Configuration > Resources > Network Interfaces**.

2 Click the **WAN1** or **WAN2** heading.
The WAN Summary panel appears.

3 Click the **IP Address** tab.
The Additional IP Address details panel appears.

4 Click the Additional IP Address you want to delete.

5 Click **Delete**.
A confirmation dialog box appears.

6 Click **Yes**.

Configuring the DLCI to IP Mapping

When connected to a Frame Relay network, BCM uses Frame Relay INARP (Inverse Address Resolution Protocol) messaging to request the next hop IP address for a given DLCI. If the other end of the connection does not support INARP messaging, there can be a communication failure because the mapping of which DLCI to use to reach a particular IP address is not known.

The DLCI to IP Mapping feature solves this problem by providing static address mapping of the DLCI to remote IP address.

Note: DLCI to IP Mapping feature is available only on the WAN1 and WAN2 interfaces.

Adding DLCI to IP Mapping

You can add up to 32 DLCI to IP Mapping entries.

To add a DLCI to IP Mapping entry

DLCI entries are added in the PVC configuration tab. The DLCI to IP mapping table allows the user to map the remote ends IP address manually to the DLCI, if the remote end does not support Inverse ARP mechanism.

1 Click **Configuration > Resources > Network Interfaces**.

2 Click the **WAN1** or **WAN2** heading.
The WAN details panel appears.

3 Click the **PVC Configuration** tab.
The PVC details panel appears.

4 Select the PVC.

5 Click the **DLCI to IP Mapping** tab.
The DLCI to IP Mapping details panel appears.

6 Enter the remote IP address for the PVC.

Chapter 61
Data modules

The following discusses configuration for modules and applications that require data or combination data/telephony line configuration.

- DDI MUX modules require two DS30 bus positions. This module supports combinations of data channels and T1 lines. A DTM (digital trunk module) contained within the module is programmed using normal DTM line configuration. The data portion of the lines are configured under the second DS30 bus as data lines. The DDI MUX supports using either the internal router or an external router. Refer to “Configuring the DDI Mux module” on page 519.
- Data modules used to configure channels on a WAN interface are always configured under DS30 Bus 08. Refer to “Configuring a data module” on page 526.

Configuring the DDI Mux module

The Digital Drop and Insert (DDI) Mux media bay module enables a BCM system to share its connection to a Universal T1 network with a local area network (LAN). A DDI Mux allows you to make more efficient use of your digital network resources and reduces the amount of equipment needed to support your voice and data networks. This module is currently available only for North American installations.

Refer to the following topics for additional information:

- “Configuring DDI Mux connections” on page 522
- “Assigning the DDI Mux modules” on page 522
- “Assigning lines for voice traffic” on page 523
- “Assigning lines to the data module” on page 523
- “Configuring the DDI Mux to work with the DTE” on page 524

DDI Mux features

The DDI Mux performs the following services:

- provides the functionality of a DTM media bay module (T1 digital lines only)
- splits the incoming T1 line so that some of the lines are used for voice traffic and some of the lines are used for data traffic
- provides either the CSU (Channel Service Unit) or DSU (Data Service Unit) functionality to support connections to data terminal equipment (DTE), such as a router or a bridge
- connects to network devices that support V.35 interfaces
- provides end-to-end transparent bit service

- supports loopbacks between the DDI Mux and the internal BCM components, and between the DDI Mux and digital terminal equipment

Note: The DTE cable that connects the BCM to the router is ordered separately from the module. If you do not have this cable, ask your customer service representative about how to obtain one.

The following figures provide examples that use internal and external routers with the DDI Mux.

Figure 158 Network overview: DDI MUX connected to 1000 hardware internal router

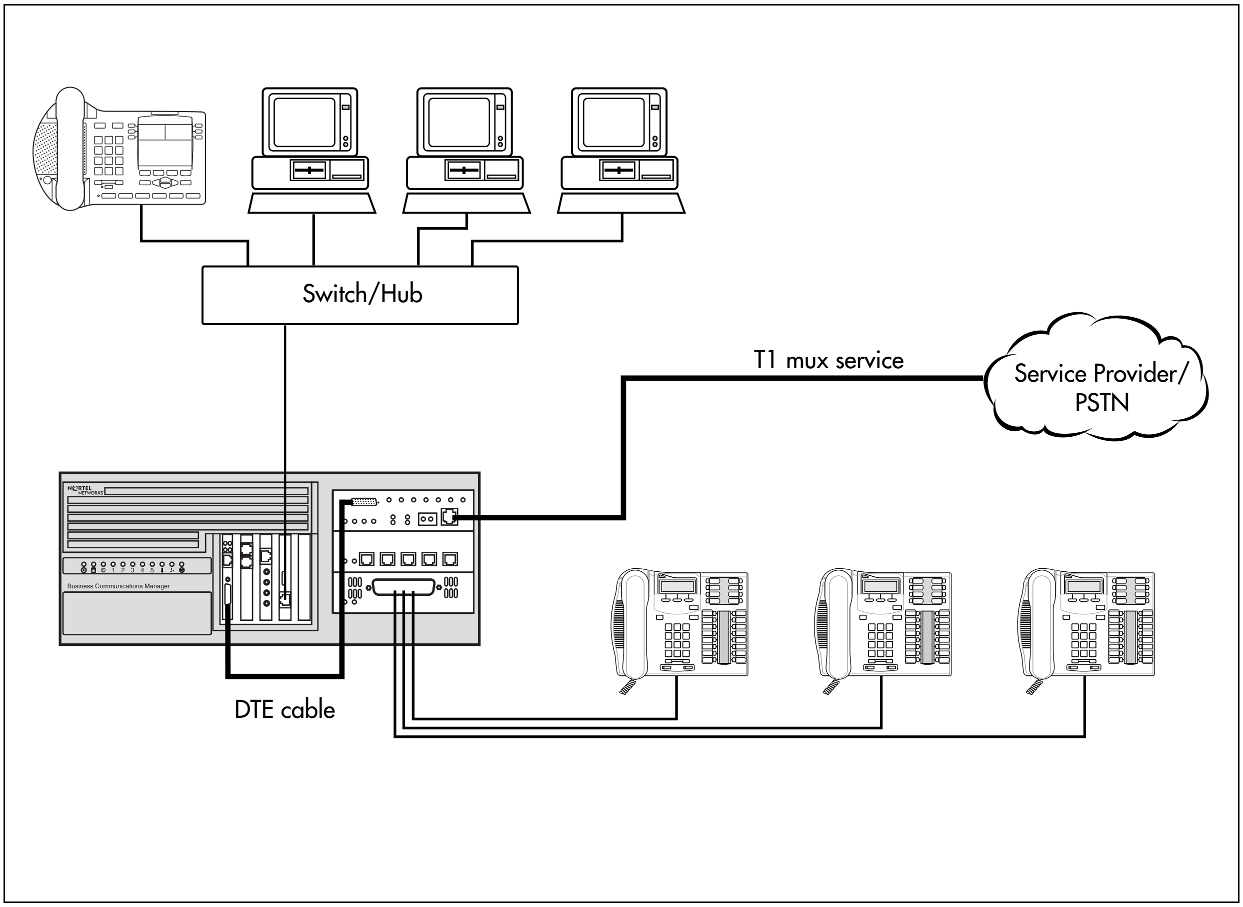


Figure 159 Network overview: DDI MUX connected to BCM400 internal router

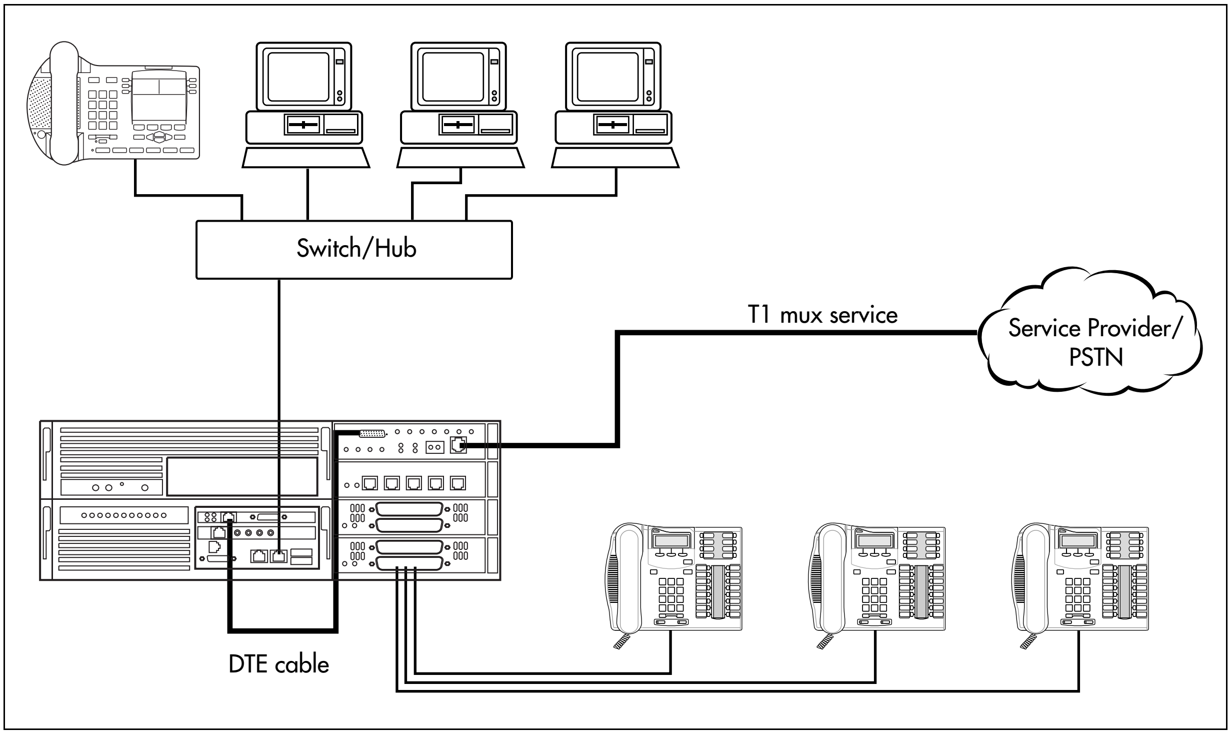
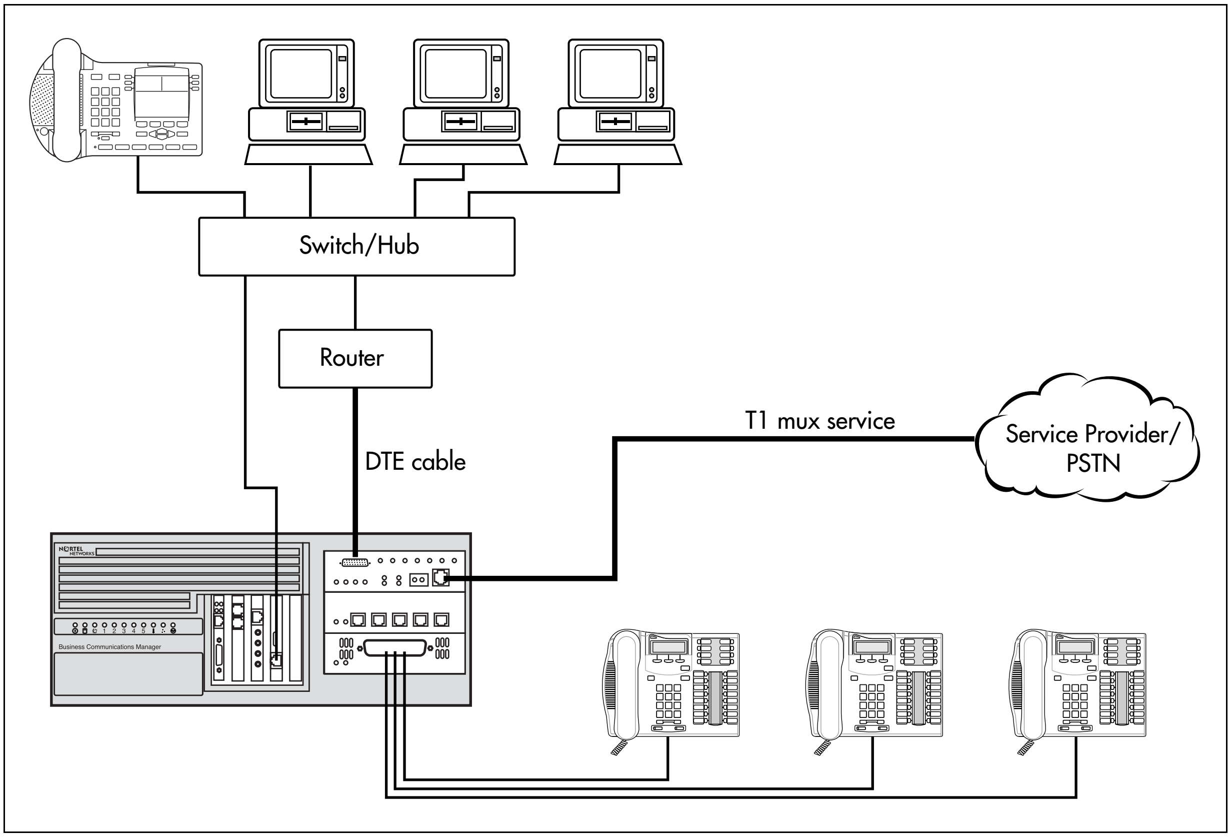


Figure 160 Overview of network using DDI Mux module with an external router



Configuring DDI Mux connections

After you install the DDI Mux, configure the module settings in the Element Manager.

To configure the DDI Mux

- assign the DDI Mux modules under **Configuration > Resources > Telephony Resources**
- assign the lines for voice traffic under **Configuration > Resources > Telephony Resources**
- assign lines to the Data Module portion of the module
- configure the DDI Mux to work with the DTE

Before you start, record the settings for the DDI Mux in the form provided in the Programming Records. At the very least, you need the following configuration information:

Table 121 Configuring DDI Mux connections

Protocol	V.35 ❑	Loopback state:	**Off** ❑ Manual DTE ❑ Manual DS30 ❑ Automatic DTE ❑
Fixed Access			
	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	
	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	
	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	
	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	Line ____ Channel ____ Line ____ Channel ____ Line ____ Channel ____	

Assigning the DDI Mux modules

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the DS30 Bus heading that was assigned to the DDI Mux.
   This Bus is the same as the DS30 bus you assigned to the DDI Mux using the DIP switches on the module.
3. Make sure the option in the **Prog Type** drop list is **Trunk module**.
4. Click the heading of the next DS30 Bus.
   This Bus is the DS30 bus automatically assigned to the Data Module portion of the DDI Mux.
5. Double-click the **Prog Type** field to activate the drop list, then click **Data module**.

Assigning lines for voice traffic

A digital T1 line has up to 24 telephone lines available for use. On the DDI Mux, you can assign some of these lines to telephony traffic and some to data traffic.

For the lines that you want to use for telephony traffic, configure the lines in the same manner as you configure lines for a DTM. For information about how to configure digital lines on a DTM, refer to "Provisioning module lines/loops" on page 130.

Assigning lines to the data module

The number of lines you assign to the data module determines the bandwidth of your data networking connection. The following table shows the allocated bandwidth for the DDI Mux according to the number of lines assigned. The allocated bandwidth also depends on the line data rate indicated by the B-channel data rate parameter.

Table 122 List of all the multiples of 56000 and 64000 bits/s

Number of lines selected	56000 bits/s	64000 bits/s
1	56000	64000
2	112000	128000
3	168000	192000
4	224000	256000
5	280000	320000
6	336000	384000
7	392000	448000
8	448000	512000
9	504000	576000
10	560000	640000
11	616000	704000
12	672000	768000
13	728000	832000
14	784000	896000
15	840000	960000
16	896000	1024000
17	952000	1088000
18	1008000	1152000
19	1064000	1216000
20	1120000	1280000
21	1176000	1344000
22	1232000	1408000
23	1288000	1472000
24	1344000	1536000

To assign a line to the data module, you must change the line type to Fixed Data Channel and then assign the line to the data module.

Assigning the line

1. In Element Manager, click **Configuration > Resources** > **Telephony Resources**.
2. Click the Bus number assigned to the Data Module.
   The details panel appears.
3. In the details panel, click the **Data module Fixed Line Assignment** tab.
   The Assigned Lines table appears.

4 Click **Add**.
The **Add Lines Assignment** dialog appears.

➡ **Note:** You can add all of the lines to the single Interface, or you can the lines to multiple interfaces.

5 In the **Add Lines Assignment** dialog box, enter the line number of one of the line assigned to the Data Module.

6 Click **OK**.

➡ **Note:** You can assign up to 24 lines to the DDI Mux.
You can assign only Fixed lines.

7 Click **Save**.

8 If you are adding all of the lines to a single Interface, repeat steps 4 to 7 for each line you want to assign to the Data Module.
If you are adding the lines to multiple Interfaces, repeat steps 4 to 7 for each line you want to assign to the Data Module.

Removing a line assignment

If you decide you want to remove a line assignment from the Data Module and use it as a telephony line, use the procedure below:

To remove a line assignment

1 Click **Configuration > Resources > Telephony Resources**.

2 Click the Bus number assigned to the Data Module.
The details panel appears.

3 In the details panel, click the **Data module Fixed Line Assignment** tab.
The Assigned Lines table appears.

4 Click on the line number you want to remove.

5 Click **Delete**.

6 On the confirmation dialog box, click **Yes**.

Configuring the DDI Mux to work with the DTE

After you have assigned the lines to the data module, you need to configure the DDI Mux so it can work with the DTE.

1 Click **Configuration > Resources > Telephony Resources**.

2 Click the Bus number for the Data Module portion of the DDI Mux.
The details panel appears.

3 Configure the setting to match the DTE you are connecting to the DDI Mux.

Table 123 describes the Data Module configuration settings.

Table 123 DDI Mux Configuration settings

Setting	Value	Description
SDI Configuration		
Protocol	V.35 RS232 RS449 EIA-530 EIA-530A	Interface standard used by the connection between the DDI Mux and the BCM WAN card or by the connection between the DDI Mux and an external router. Default: V.35
B channel data rate	64 K 56 K	Select the transmission rate per channel (Fixed data lines) assigned to the module.
Transmit clock source	Auto DCE DTE	The DDI Mux requires a timing signal to clock data transmitted by the DTE. This signal can be supplied by the DTE or generated internally by the BCM. If the signal is generated by the DTE, the clock must be locked to the frequency of the DDI Mux clock. • **Auto**: The DDI Mux checks if the clock provided by the DTE is valid. If the clock is valid, the module uses DTE clocking. If the DTE clocking is not valid, the module uses its own internal clock (DCE). • **DCE**: The internal TxSync clock is used to clock data. • **DTE**: The TXClk clock provided by the DTE is used by the DDI Mux to clock data. **Note**: Use the DTE option only for diagnostic purposes. For all options, the DTE must synchronize to the module.
Transmit clock inversion	<check box>	When the internal DCE signal is used to clock in data, signal delays caused by cable length can cause clocking errors. To adjust for round trip delays between the DDI Mux and DTE, invert the internal clock used by the module to clock in data from the DTE. To enable clock inversion, select the **DTE** clock or **Auto** clock.
Data inversion	<check box>	When Data Inversion is selected, the DDI Mux inverts the data before routing it to the T1 connection and DTE. Inversion allows the module to use the properties of protocols such as HDLC/SDLC (a transmission standard for data) to meet the ones density requirement of the network. This feature must be available and activated at the far end.
Loopback status		
Loopback state	<read-only>	Indicates the loopback state.
Loopback	Off Manual DTE Manual DS30 Automatic DTE	Off: No loopback is active. Manual DTE: The SDI takes the data received from the DTE and loops it back towards the DTE. Manual DS30: The SDI takes the data received from the DS30 bus and loops it back. Automatic DTE: The SDI enters in the loopback state when requested by the DTE. Auto DTE should only be set when supported by DTE.

Configuring a data module

DS30 bus 08 is reserved for configuring circuit switched B-channels as a WAN interface for the BCM. This allows the Integrated QoS Routing feature to create one or more dial up ISDN connections via the PSTN network using PRI or BRI trunks. BCM automatically configures the Module type as a Data Module and sets the Data module type to Baystack. Baystack is the only setting supported on BCM.

The following includes information about:

- "Viewing the data module settings"
- "Programming the BayStack settings"

Viewing the data module settings

To view the current settings for the data module

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.

Programming the BayStack settings

When you select the BayStack data module, the following detail configuration tabs appear:

- Data Module Parameters
- Data Module Line Assignment
- Data Module Line Pool Access

Refer to the following information:

- "Fixed access" on page 526
- "Switched access (PRI & BRI)" on page 527
- "Line assignment" on page 528
- "Line pool access" on page 529

Fixed access

Fixed access is supported for the Norstar Data Interface (NDI) only.

To assign one or more Fixed lines to the data module

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.
3. In the details panel, click the **Data Module Fixed Line Assignment** tab.
   The Assigned Lines table appears.
4. Click **Add**.
   The **Add Lines Assignment** dialog appears.
5. In the **Add Lines Assignment** dialog box, enter the line number of one of the fixed line assigned to the interface.
6. Click **OK**.
7. Click **Save**.
8. If you are adding all of the lines to a single Interface, repeat steps 4 to 7 for each line you want to assign to the Data Module.
   If you are adding the lines to multiple Interfaces, repeat steps 4 to 7 for each line you want to assign to the Data Module.
9. Select **Unassigned** or **Assigned**.
10. Record each channel and line combination. Each channel used by the BayStack data module maps to a line.

To delete a line assignment

1. Click **Configuration** > **Resources** > **Telephony Resources**.
2. Click the Bus number assigned to the Data Module.
   The details panel appears.
3. In the details panel, click the **Data module Line Assignment** tab.
   The Assigned Lines table appears.
4. Click on the line number you want to remove.
5. Click **Delete**.
6. Click **Yes**.

Switched access (PRI & BRI)

You can assign ISDN lines to the BayStack data module to provide:

- normal data network access for the data module
- dial-up backup and overflow bandwidth (additional channels or trunks) as needed

Line assignment

You can assign one or more lines to the BayStack data module for incoming data transmission.

Note: The data module will answer data calls only. It will not answer voice calls.

Adding line assignments

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.
3. Click the **Data Module Line assignment** tab.
4. Click **Add**.
   The Add Line assignment panel appears.
5. In **Line** box, enter the number of the trunk or a target line you need to assign to the BayStack data module.
6. Click **Save**.
7. Click the heading for the line you added.
8. In the **Dial-in number** box, enter the Dial-In Number for the line (up to 24 digits). The number must match the Dial-In Number entered for the line and channel in BayStack data module programming.
9. Repeat steps 4 to 8 to assign additional lines to the BayStack data module as required.

Deleting line assignments

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.
3. Click the **Data Module Line assignment** tab.
4. Click on the line number you want to remove.
5. Click **Delete**.
   A confirmation dialog box appears.
6. Click **Yes**.

Line pool access

You can give the BayStack data module access to a line pool for outgoing data transmission.

Adding line pool access

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.
3. Click the **Line pool access** heading
4. Click **Add**.
   Or, right click the **Line pool access** heading and click **Add**.
   The Add Line pool access panel appears.
5. In **Pool** box, enter the letter of the line pool to provide access to the BayStack data module. You must program line pool access when you select the switched access settings for the BayStack data module. To use PRI line pools, program the BayStack data module to use a destination code.
6. Click **Save**.

Deleting line pool access

1. Click **Configuration > Resources > Telephony Resources**.
2. Click the Bus number 8.
   The details panel appears.
3. Click the **Data Module Line Pool Access** tab.
4. Click the line pool you want to delete.
5. Click **Delete**.
   A confirmation dialog box appears.
6. Click **Yes**.

Chapter 62

Configuring the Dial-up resources

BCM allows the creation and use of dial-up connections for the following functionalities:

Note: Modem interfaces are not listed as default interfaces, the administrator must add them.

Dial-in

Dial-in functionality allows the administrator to access the BCM remotely by making an IP connection using an ISDN PRI/BRI line, or a modem, and also can be used for various networking applications.

When the System Administrator connects to the BCM they can access all IP-based system management operations.

In order to improve security, BCM supports callback functionality for ISDN and modem interfaces. When incoming calls are received on the interface, BCM terminates the call and makes an outgoing call to the remote end based on the callback number specified and creates a reverse connection.

Dial-out

Dial-out for Primary WAN Connection

Dial-up links can be used as primary connections to external networks. The modem provides the functionality of dialing out to the ISP over a normal telephone line and establishing an IP link. This can serve as the primary connection through which private networks behind BCM can access the Internet.

PPPoE also provides this functionality but is used when connecting to broadband modems. PPP connection is established over Ethernet medium. When BCM uses a PPPoE connection, the ISP can control access billing and other types of service on a per-user basis rather than on a per-site basis.

ISDN PRI/BRI lines can be configured for primary connections when ISP supports ISDN connectivity. The BCM dials out over the ISDN channels configured on the device and establishes the PPP connection.

Dial-out for WANBackup

Modem, ISDN and PPPoE interfaces can be configured as back-up connections for permanent WAN connections. To configure an interface as a backup connection it must first be enabled, then if the BCM experiences a primary connection failure, it will dial-out using the dial-up interface configured as the backup. The connection is established over this link and the connectivity to the external network is restored. Whenever the primary connection is available again, BCM disconnects the dial-out connection. This process helps to maintain high availability of networks, and minimal downtime. Only one interface can be configured as back-up at any point of time.

Dial-out for SNMP traps

All dial-up interfaces can be configured to send SNMP traps to a remote Management Station. Whenever BCM needs to send SNMP traps, it dials out through the dial-up interface that is configured for SNMP traps. The SNMP traps are sent over the IP connection using the dial-up connection. After sending the traps, the dial-up connection is disconnected after a specified time interval during which there has been no traffic over the link.

Dial-out for Scheduled Backup, Scheduled Log Transfer, Scheduled Software Update

Modem and ISDN interfaces can be used for sending scheduled backups of configuration data, scheduled backups of log files collected on BCM and for scheduled software updates on BCM. When the timer for these scheduled activities expires, a dial-out connection is initiated over the dial-up interfaces.

The system administrator can manually initiate dial-out.

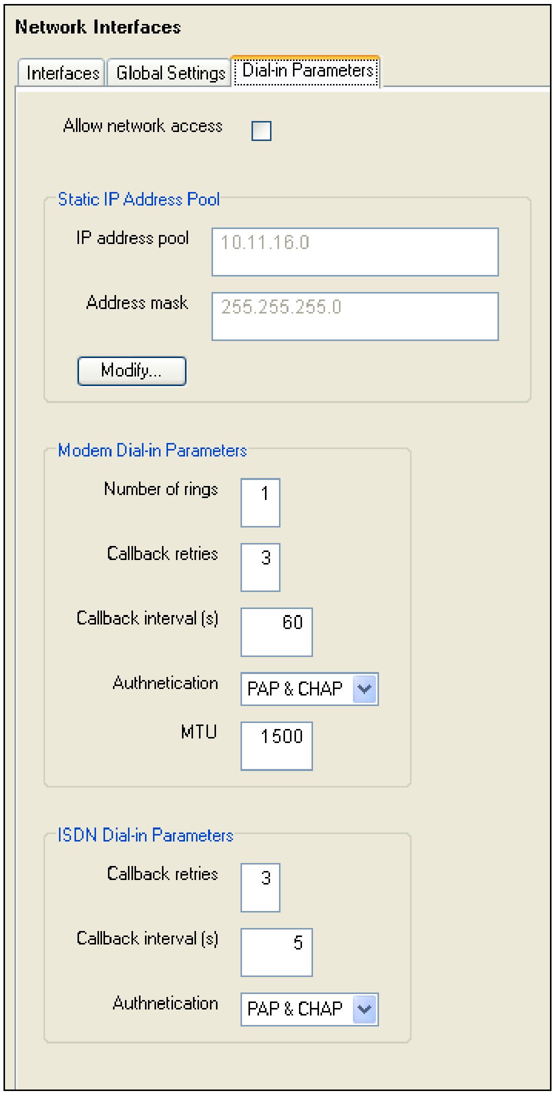
Refer to the following topics for additional information:

- “Configuring the Dial-in Parameters”
- “Configuring a modem interface” on page 537
- “Configuring an ISDN interface” on page 543
- “Configuring a PPPoE interface” on page 550
- “Creating an auto dial-out interface” on page 557
- “Guidelines for using remote Dial-in” on page 557
- “Using a dial-up interface as a primary connection” on page 557

Configuring the Dial-in Parameters

→ **Note:** The default IP address Pool must be changed on one end of a dial-up connection between two BCMs.

Figure 161 Dial-in Parameters



Allow network access □

Static IP Address Pool

IP address pool 10.11.16.0

Address mask 255.255.255.0

Modem Dial-in Parameters

Number of rings 1

Callback retries 3

Callback interval (s) 60

Authnetication PAP & CHAP

MTU 1500

ISDN Dial-in Parameters

Callback retries 3

Callback interval (s) 5

Authnetication PAP & CHAP

Table 20 Dial-in Parameters panel fields (Sheet 1 of 3)

Attribute	Value	Description
Allow network access	<check box>	Select whether to allow dial up access to the entire network. When using dial up for dial-on-demand WAN connection (as a primary or back up WAN connection), select the Allow Network Access check box. When using dial up for remote system management purposes only, clear the Allow Network Access check box.
Static IP Address Pool		

Table 20 Dial-in Parameters panel fields (Sheet 2 of 3)

Attribute	Value	Description
IP address pool	<IP address>	Enter the IP address BCM assigns when a remote site dials into the BCM system. The address mask determines the IP address range. These IP addresses are used for assigning to remote sites as well as to the BCM local interface. For example, if the static IP address is 10.11.16.0 and the address mask is 255.255.255.240, this gives an address range of 10.11.16.1 to 10.11.16.15. The first IP address is assigned to the modem interface i.e. 10.11.16.1 and 10.11.1.2, one for the local modem interface and the other for the remote assignment. Subsequent IP addresses are assigned to the ISDN interfaces, one for the local interface and the other for the remote assignment. **Note:** If you have enabled Web Cache, you must update the Web Cache page when you change the IP address of the interface. Refer to "Web Cache page update required" on page 686. Default: 10.11.16.0
Address mask	<IP address>	Enter the IP address mask corresponding to the IP address range. The IP addresses from the static address pool then reserved for assignment to remote sites. Default: 255.255.255.0
Actions		
Modify	<button>	Click to modify the dial-in parameter details.
Modem Dial-in Parameters		
Number of rings	<1-20>	The number of rings the BCM will wait before accepting the call. Default: 1.
Callback retries	<1-10>	This parameter is the number of attempts made by BCM to dialout to the remote end during callback. Default: 3.
Callback intervals(s)	<0-65536>	Interval for successive connection attempts for dialout during callback. Default: 60 seconds.
Authentication	PAP CHAP PAP & CHAP	Specify PAP or CHAP to force the far end to conform to that protocol. When you select PAP & CHAP you allow the far end to negotiate which protocol to use. Default: PAP & CHAP
MTU	<128-16384>	Maximum size of the packets that will be sent. Default: 1500.
ISDN Dial-in Parameters		
Callback retries	<0–10>	This parameter is the number of attempts made by BCM to dialout to the remote end during callback. Default: 3.

Table 20 Dial-in Parameters panel fields (Sheet 3 of 3)

Attribute	Value	Description
Callback interval (s)	<0 – 65536>	Interval in seconds for successive connection attempts for dialout during callback. Default: 60.
Authentication	PAP CHAP PAP & CHAP	When you specify PAP or CHAP on your system you are forcing the far end to conform to that protocol. When you select PAP & CHAP you allow the far end to negotiate which protocol to use. Default: PAP & CHAP

To access the BCM for maintenance over an analog line

1. Connect an ATA2 to the BCM modem.
2. Under the **Modem Dial-In Parameters** panel set the **Number of rings** to 4, or greater.
3. Dial-in to the BCM operator.
4. The operator will recognize the assigned tone of the incoming call and transfer the call to the ATA2 DN.

Note: If a site experiences problems connecting to the ATA2 via the modem connection, temporarily disable Music On Hold (MOH) under **Configuration > Telephony > Feature Settings**, if applicable, and try connecting again.

To configure the Dial-in Parameters

1. Click **Configuration > Resources > Network Interfaces**.
2. Select the device to configure.
3. Click the **Dial-in Parameters** tab.
4. Configure the RAS Server TCP/IP settings. Refer to the information in Table 20.

Modem Dial-in Parameters

- BCM is equipped with an internal modem that connects to your phone line with a RJ-11 connector. The V.92 modem has the following features: V.92 56 kbps ITU standard
- V.34 48 kbps ITU standard
- V.42/MNP 2-4 error control
- V.42 bis/MNP 5 data compression
- compatible with ITU and Bell Standards from 56 kbps down to 1200 bps

➡ **Note:** The modem is capable of receiving at a maximum speed of 56 kbps and transmitting at a maximum speed of 31.2 kbps. Because of FCC regulations, receiving speed is limited to 53 kbps. Current line noise can impact the speed of the modem.

The V.92 modem WAN connection always uses PPP as the link layer protocol. For correct operation, the link must be connected to a remote access server (RAS).

BCM supports the following authentication protocols for PPP:

- Password Authentication Protocol (PAP)
- Challenge Authentication Protocol (CHAP)

Users must be assigned dial-in privileges to use the Modem Dialup. Dial-in privileges are assigned to users by adding them to Remote Access group in Element Manager **Configuration > Administrator Access > Accounts and Privileges >View by Groups > Members** tab (subpanel).

Refer to the following topics for additional information:

- "Creating a modem interface"
- "Enabling and disabling the modem interface"
- "Configuring a modem interface" on page 537

Creating a modem interface

➡ **Note:** Modem interfaces are not listed as default interfaces, the administrator must add them.

To create a modem interface

1. Click **Configuration > Network Interfaces > Interfaces** tab.
2. Click **Add**.
   The **Add Interface** dialog box appears.
3. Select **Modem** from the **Interface type** drop-down list.
4. Enter a logical name in the **Interface name** field.
5. Click **OK**.
   The modem appears in the list of Network Interfaces.

Enabling and disabling the modem interface

If you want to the use the modem as a backup WAN connection or as interface to send SNMP traps to the SNMP Manager, you must enable the modem interface.

If you are not using the modem interface for WAN backup or SNMP traps, Nortel recommends that you disable the modem to help prevent unauthorized access to the BCM.

To enable or disable the modem interface

1 Click **Configuration > Resources > Network Interfaces** and select Modem from the Interfaces table.

2 Select to enable, or clear to disable, the check box that corresponds to the selected modem in the **Enable** column.

Changing the Modem Region

There are several internal modem settings that vary depending on the country in which the modem is operating. BCM uses the country you select in the Modem Region parameter to properly configure these internal settings.

Normally, the Modem Region is set using the Startup Profile. However, in situations where the Modem Region was not set using the Startup Profile, you can use Element Manager to set the Modem Region.

To change the Modem Region

1 Click **Configuration > Resources > Network Interfaces > Global Settings** tab.

2 Click the **Modem Region** drop-down list and click the country in which the BCM system resides.

Configuring a modem interface

The modem is used as a dial up connection and as a backup connection.

➡ **Note:** Remember to set Dial Up global parameters before creating modem dial up interfaces. For information about setting Dial Up global parameters see "Configuring the Dial-in Parameters" on page 532

The same modem may be shared between the remote dial-in for system administration and the backup WAN link. The WAN backup function is not available if a break in the WAN permanent connection occurs while a system administrator is connected to BCM using the modem.

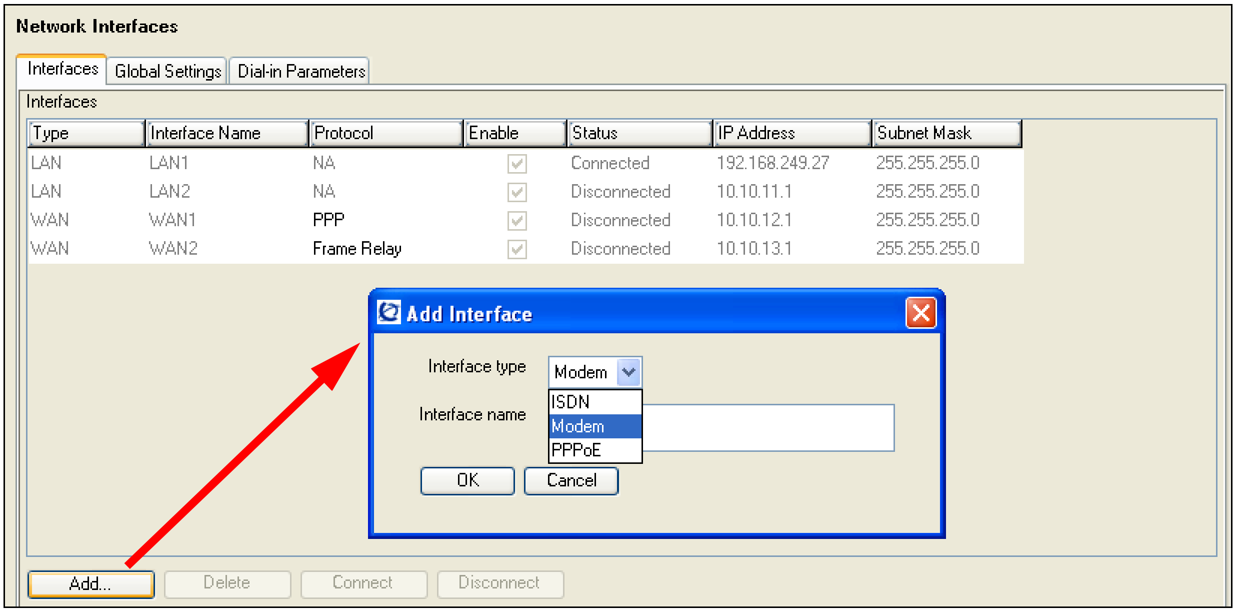
To configure the modem as a backup interface

1 Click **Configuration > Resources > Network Interfaces > Interfaces tab** and click **Modem**.

2 Click the **Modem** heading to configure as a backup connection.

3 Configure the modem. Refer to the information in Table 125, Table 126, and Table 127.

Figure 162 Interface summary panel



Network Interfaces

Interfaces | Global Settings | Dial-in Parameters

Interfaces

Type	Interface Name	Protocol	Enable	Status	IP Address	Subnet Mask
LAN	LAN1	NA	☑	Connected	192.168.249.27	255.255.255.0
LAN	LAN2	NA	☑	Disconnected	10.10.11.1	255.255.255.0
WAN	WAN1	PPP	☑	Disconnected	10.10.12.1	255.255.255.0
WAN	WAN2	Frame Relay	☑	Disconnected	10.10.13.1	255.255.255.0

Add Interface

Interface type: Modem

Interface name:

Table 124 Interface summary parameters

Attribute	Value	Description
Type	<read-only>	Displays the type of interface.
Interface Name	<alphanumeric>	Enter a name for the modem interface.
Protocol	<read-only>	Only PPP is supported on dial-up links.
Enable	<check box>	Select to enable the modem dial-out interface.
Status	Connected Disconnected	View the modem interface status and connect or disconnect the modem connection. The possible states are: Connected: The modem is enabled and the dial-up link is currently active. Disconnected: The modem interface is enabled and the dial-up link is currently disconnected.
IP Address	<IP address>	Enter the IP address of the modem interface when it connects. Users can set a fixed IP address for the dial-up interface. If a fixed address is specified, BCM uses this IP address on the receiving end. Users can select RemoteAssigned to indicate that BCM must obtain an IP address from the remote end and use it. The address obtained depends on the RAS server to which BCM connects. Default: Remote Assigned.
Subnet Mask	<IP address>	Net mask of the interface. In case of dialup connections, the mask value will be a read-only parameter. Default: 255.255.255.255

Link Parameters

To configure the Modem Link Parameters tab

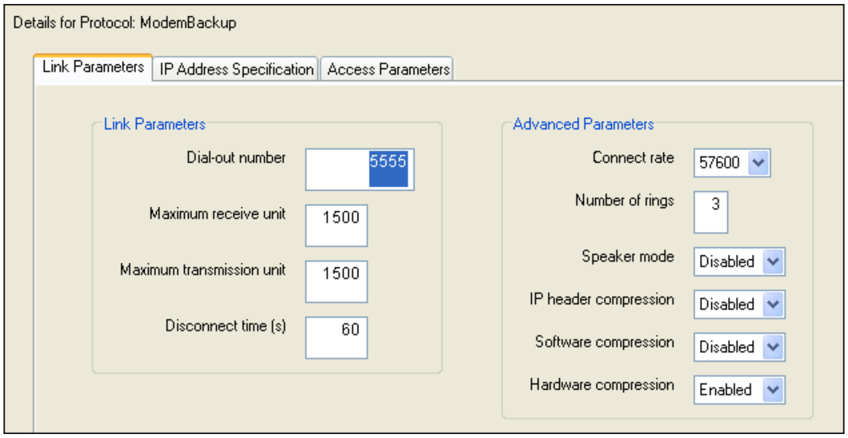
1 Click **Configuration > Resources > Network Interfaces > Interfaces** tab.

2 Click the Modem interface to configure.

3 Click the **Link Parameters** tab.
The Link Parameters panel appears. See Figure 163.

4 Configure the Modem Link Parameters. Refer to the information in Table 125.

Figure 163 Link Parameters



Details for Protocol: ModemBackup

Link Parameters

Dial-out number: 5555

Maximum receive unit: 1500

Maximum transmission unit: 1500

Disconnect time (s): 60

Advanced Parameters

Connect rate: 57600

Number of rings: 3

Speaker mode: Disabled

IP header compression: Disabled

Software compression: Disabled

Hardware compression: Enabled

Table 125 Modem link parameters (Sheet 1 of 2)

Attribute	Value	Description
Link Parameters		
Dial-out number	<numeric>	Type a telephone number to use to connect using the modem interface. If needed, include area codes and all necessary digits to dial an external number.
Maximum receive unit	<128-16384>	The largest incoming packet size the modem will handle. Default: 1500.
Maximum transmission unit	<128-16384>	Maximum size of the packets that will be sent. Default: 1500.
Disconnect time (s)	<numeric>	Enter the interval, in seconds, during which the modem interface disconnects when there is no traffic. Select **0** if you want the BCM to close the connection when the far end hangs up. Specifying a value of 0 makes the connection persistent. **Note**: If you have more than one Trap Community Entry configured with this dial up interface, using a very small number for the Disconnect Time will put you at risk of a racing condition. To reduce the racing condition, enter a larger value for the Disconnect Time.
Advanced Parameters		
Connect rate	57600 38400 19200 9600 4800	Specify the initial speed (in bits per second) for the modem to connect. Set to the maximum permissible value for best results. **Note**: This is the initial rate; the actual rate is always negotiated. Default: 57600

Table 125 Modem link parameters (Sheet 2 of 2)

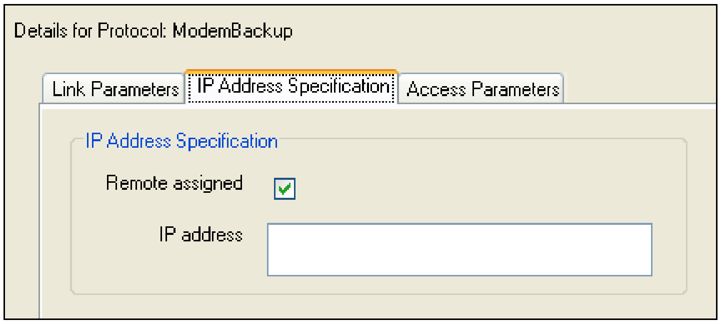
Attribute	**Value**	**Description**
Number of rings	<numeric>	Specify the number of rings the BCM waits before determining that the far end of the connection is not answering. Default: 3
Speaker mode	Enabled Disabled	Enable or disable the speaker during initial link establishment. Default: Disabled
IP header compression	Enabled Disabled	Enable or disable IP header compression. To function, the receiving end must also use this feature. Default: Disabled
Software compression	Enabled Disabled	Enable or disable data compression in the software, instead of the modem. For dial-up connections, Element Manager uses Microsoft Point-to-Point Compression algorithm (MPPC). Default: Disabled
Hardware compression	Enabled Disabled	Enable or disable data compression in the hardware instead of the software. Default: Enabled

IP Address Specification

To configure the Modem IP Address Specification tab

1. Click **Configuration > Resources > Network Interfaces > Interfaces** tab.
2. Click the Modem interface to configure.
3. Click the IP Address Specification tab.
   The IP Address Specification panel appears. See Figure 164.
4. Configure the IP Address Parameters. Refer to the information in Table 126.

Figure 164 IP Address Specification panel



Details for Protocol: ModemBackup

Remote assigned ☑

IP address

Table 126 IP Address Specification fields

Attribute	Value	Description
IP Address Specification		
Remote assigned	<check box>	Selected: The BCM obtains IP address from the remote end. Cleared: A static IP address needs to be configured in the IP address Default: Selected
IP address	<IP address>	Enter a static IP address when the Remote Assigned check box is cleared.

Access Parameters

To configure the Modem Access Parameters tab

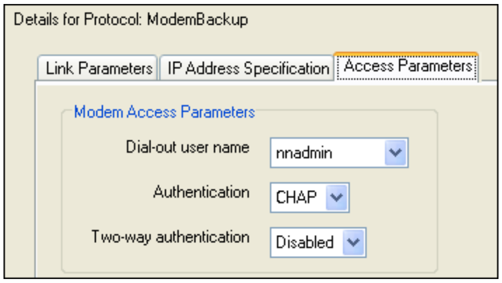
1 Click **Configuration > Resources > Network Interfaces > Interfaces** tab.

1 Click the Modem interface to configure.

2 Click the **Access Parameters** tab.
The Modem Access Parameters details panel appears. See Figure 165.

3 Configure the Access Parameters. Refer to the information in Table 127.

Figure 165 Access Parameters



Details for Protocol: ModemBackup

Modem Access Parameters

Dial-out user name: nnadmin

Authentication: CHAP

Two-way authentication: Disabled

Table 127 Access parameters

Attribute	**Value**	**Description**
Modem Access Parameters		
Dial-out user name	nnadmin BTRemoteIsdn	Select a user name that the link uses to authenticate itself when dialing out to another router. Default: nnadmin
Authentication	CHAP PAP	Select authentication type for the link. CHAP is used first and if the receiving end of the link declines. PAP is used to authenticate the link.
Two-way Authentication	Enabled Disabled	Enable or disable link authentication in both directions. Default: Disabled.

BCM supports ISDN dial up for dial-on-demand WAN access. You have the choice to use ISDN BRI/PRI as a persistent or dial-on-demand WAN connection or as a backup for your permanent WAN connection.

Note: To use an ISDN dial-up connection, you must first configure your system for ISDN. For more information, refer to "ISDN overview" on page 701. If your system is already configured to support ISDN, make sure you configure a Data Module for ISDN dial up connection. For more information, see "Configuring a data module" on page 526.

After you have created an ISDN dial up interface, you must use "Configuring Net Link Manager" on page 561 to select which type of network connection the system must use for primary and backup connection.

Refer to the following topics for information:

- "Creating an ISDN dial out interface"
- "Configuring an ISDN interface" on page 543
- "ISDN Channel Characteristics" on page 544
- "To delete an ISDN interface" on page 548

Creating an ISDN dial out interface

To create an ISDN dial out interface

1 Click **Configuration** > **Resources** > **Network Interfaces**.

2 Click **Add**.
The **Add Interface** dialog box appears.

3 Select **ISDN** from the drop-down list.

4 In the **Interface Name** box, type the name of the interface you are creating.
This is the logical name for the interface and is not related to the Dial In username. The Dial-in username is added under **Configuration > Administrator Access > Accounts and Privileges.**

Caution:
If you are creating an ISDN interface to use as a backup for a permanent WAN connection, the **Interface name** must contain the string "backup". For example, "ISDNbackup" is a valid name if you want to use an ISDN connection as a WAN backup connection.

Configuring an ISDN interface

To configure an ISDN interface

1 Click **Configuration** > **Resources** > **Network Interfaces**.

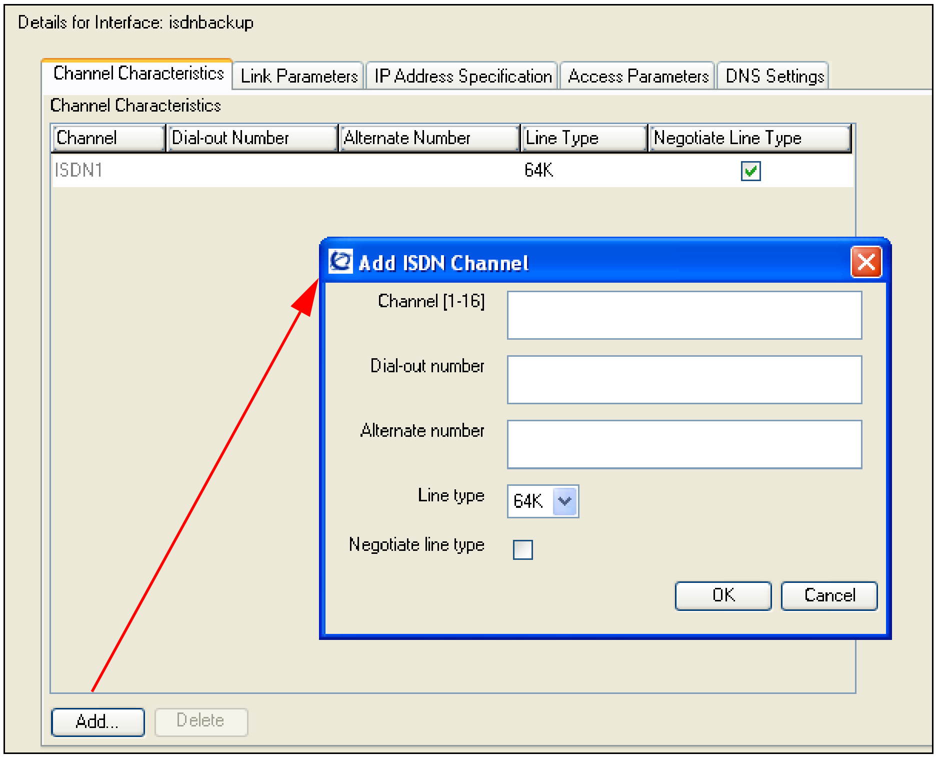
2 Click on the ISDN interface to configure.
The ISDN details panel appears.

3 Configure the ISDN device. Refer to the information in Table 21.

Note: You cannot select an ISDN interface that is set to "Remote assigned" as the Local Gateway IP for the VoIP Gateway.

ISDN Channel Characteristics

Figure 166 ISDN Channel Characteristics



Details for Interface: isdnbackup

Channel Characteristics

Channel	Dial-out Number	Alternate Number	Line Type	Negotiate Line Type
ISDN1			64K	☑

Channel [1-16]

Dial-out number

Alternate number

Line type 64K

Negotiate line type □

Table 21 ISDN Channel Characteristic fields

Attribute	Value	Description
Channel	<1-16>	Select one or more of the ISDN channels to be used for dialout. There are 16 ISDN ports available, named ISDN1 to ISDN16.
Dial-out number	<numeric>	Enter the primary phone number to use to make an ISDN connection. If needed, include area codes and all necessary digits to dial an external number. The phone number must contain only numerical digits (no alphabetical or other characters are allowed). Default: Blank.
Alternate number	<numeric>	Alternate Phone number in case the Primary Phone number is unreachable. Default: Blank
Line type	56K 64K	Select either a 64K Digital or 56K Digital line. BCM ISDN supports two types of Unrestricted Digital Information (UDI) bit streams: UDI, and UDI-56. With UDI, data is transmitted at 64kbps (64K Digital). With UDI-56, a 1 bit is inserted in the eighth bit position of each B-channel time slot while the other 7 bits form the 56 kbps channel (56K Digital). Default: 64K Digital.
Negotiate line type	<check box>	Select whether or not the system will select a line with a slower speed if it is unable to connect at the previously set speed. Default: Enabled.

ISDN Link Parameters

To configure the ISDN Link Parameters

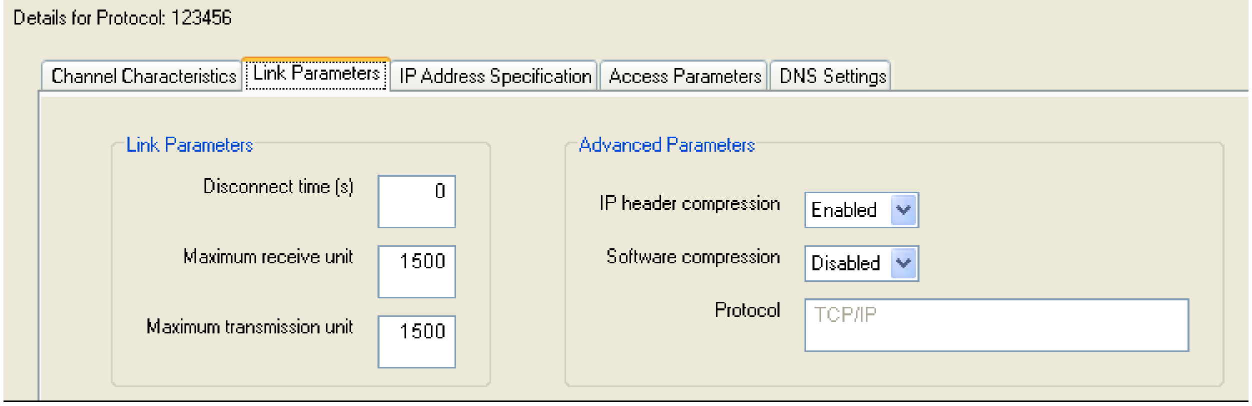
1 Click **Configuration > Resources > Network Interfaces > Interfaces** tab.

2 Click the ISDN interface to configure.

3 Click the **Link Parameters** tab.
The Link Parameters panel appears. See Figure 167.

4 Configure the ISDN Link Parameters. Refer to the information in Table 128.

Figure 167 ISDN Link Parameters


Details for Protocol: 123456

Link Parameters

Disconnect time (s): 0

Maximum receive unit: 1500

Maximum transmission unit: 1500

Advanced Parameters

IP header compression: Enabled

Software compression: Disabled

Protocol: TCP/IP

Table 128 ISDN link parameters

Attribute	**Value**	**Description**
Link Parameters		
Disconnect time (s)	Numeric	Enter the interval, in seconds, during which the ISDN interface disconnects when there is no traffic. Specifying a value of 0 makes the connection persistent. **Note:** If you have more than one Trap Community Entry configured with this dial up interface, using a very small number for the Disconnect Time will put you at risk of a racing condition. To reduce the racing condition, enter a larger value for the Disconnect Time. Default: 60
Maximum receive unit	<128 – 16384>	The maximum size of the packets that can be received. Default: 1500.
Maximum transmission unit	<128 – 16384>	The maximum size of the packets that will be sent. Default: 1500.
Advanced Parameters		
IP header compression	<drop-down list>	Enable or disable IP header compression. The feature must be enabled at both ends of the connection. Default: Enabled.
Software compression	<drop-down list>	Enable or disable software compression. Default: Disabled.
Protocol	<read-only>	TCP/IP is the protocol that this ISDN interface uses.

ISDN Access Parameters

To configure the ISDN Access Parameters

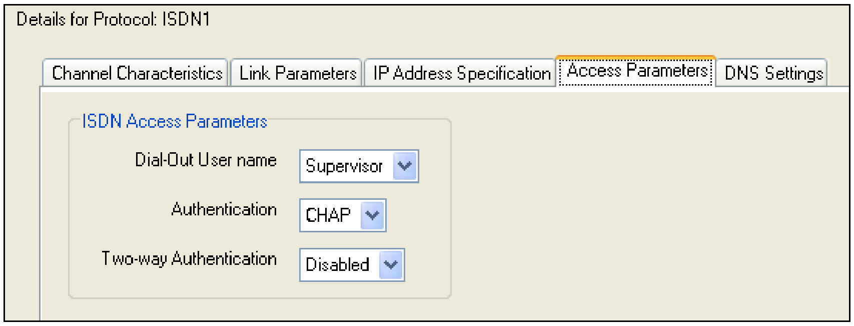
1 Click **Configuration > Resources > Network Interfaces > Interfaces** tab.

2 Click the ISDN interface to configure.

3 Click the **Access Parameters** tab.
The Access Parameters panel appears. See Figure 168.

4 Configure the ISDN Access Parameters. Refer to the information in Table 129.

Figure 168 ISDN Access Parameters



Details for Protocol: ISDN1

Dial-Out User name: Supervisor
Authentication: CHAP
Two-way Authentication: Disabled

Table 129 ISDN access parameters

Attribute	Value	Description
ISDN Access Parameters		
Dial-out user name	nndamin BTRemoteIsdn	The user name used for authenticating to the remote end. Default: nnadmin
Authentication	PAP CHAP	Select the authentication type for the link. The options are **PAP** or **CHAP**. **PAP**: When selected, the PAP is used for authentication. **CHAP**: When selected, CHAP is used for authentication. Default: CHAP.
Two-way Authentication	Enabled Disabled	Enable or disable link authentication in both directions. Default: Disabled.

Assigning an ISDN dial number and IP address

With an ISDN dial-out interface, you can bundle more than one ISDN channel to increase the throughput. This is referred to as multi-link support.

With each ISDN channel configuration, you can choose to configure a primary dial number (Phone1), and a backup dial number (Phone2). However, you are not allowed to assign different IP addresses for the different phone numbers you are dialing. The main reason behind this restriction is, with multi-link, even if you can reach your destination via different phone numbers, once your ISDN pipe is established, there is only one source and one destination from the IP-layer.

If you assign Phone1 to reach Site A and Phone2 to reach Site B, and Site A and Site B belong to different subnets, the pre-assigned IP address scheme will not work. In this scenario, you must use the Remote Assigned IP address option. For information about setting the IP Address to the RemoteAssigned, refer to “Configuring an ISDN interface” on page 543. For information about configuring Net Link Manager, refer to “Configuring Net Link Manager” on page 561.

ISDN Channel Characteristics

To modify the characteristics of an existing ISDN channel

1. Click **Configuration > Resources > Network Interfaces > Interfaces** tab.
2. Click the ISDN interface to configure.
3. Select the **Channel Characteristics** tab.
4. Double-click the field to modify.
5. Make the necessary changes.

To delete an ISDN channel from the ISDN Channel Characteristics table

1. Click **Configuration > Resources > Network Interfaces > Interfaces** tab.
2. Click the ISDN interface to modify.
3. Click the ISDN Channel Characteristic you want to delete.
4. Click **Delete.**
   A confirmation dialog box appears.
5. Click **Yes**.

To delete an ISDN interface

1. Click **Configuration > Resources > Network Interfaces**.
2. Click the ISDN interface you want to delete.
3. Click **Delete**.
   A confirmation dialog box appears.
4. Click **Yes**.

Point to Point Protocol over Ethernet (PPPoE)

Refer to the following topics:

- "Settings required for PPPoE"
- "Creating a PPPoE dial up interface" on page 549
- "Configuring a PPPoE interface" on page 550
- "Deleting a PPPoE interface" on page 555

PPPoE is the protocol BCM uses when connecting to a data network using a broadband modem. Digital Subscriber Line (DSL) modems and cable modems are examples of broadband modems.

When the BCM uses a PPPoE connection, the Internet Service Provider (ISP) can control access, billing and other types of service on a per-user, rather than a per-site basis.

Settings required for PPPoE

The data packets that pass through the PPPoE connection interact with other routing features in BCM. As a result, there are several settings you must configure in other features so those features can use the PPPoE connection. Table 130 shows the features that interact with PPPoE.

Note: To use PPPoE, you must have a BCM system that has two LAN cards.

Table 130 Features that interact with PPPoE

Feature	Description of interaction
LAN interfaces	When you install PPPoE, the LAN1 interface is dedicated to PPPoE. You must not use the LAN1 interface for any other purpose.
IPSec Tunnels	To use IPSec tunnels over the PPPoE interface, BCM requires a single known IP address be assigned to the PPPoE interface. If your Internet Service Provider uses DHCP to assign the IP addresses, the DHCP server must assign the same IP address to the PPPoE interface every time BCM connects.
Internet Clients	Clients who want to use the BCM PPPoE interface to access the internet must set their MTU size to a value less than or equal to 1480 bytes, but not less than 1400 bytes.
Software Keycode	You must purchase and install the PPPoE Software keycode before you can install PPPoE. For information about purchasing the PPPoE Software Keycode, contact your Nortel representative. For information about how to install the PPPoE Software Keycode, refer to the Software Keycode Installation Guide that comes with your Software Keycode.

Creating a PPPoE dial up interface

To create a PPPoE dial up interface

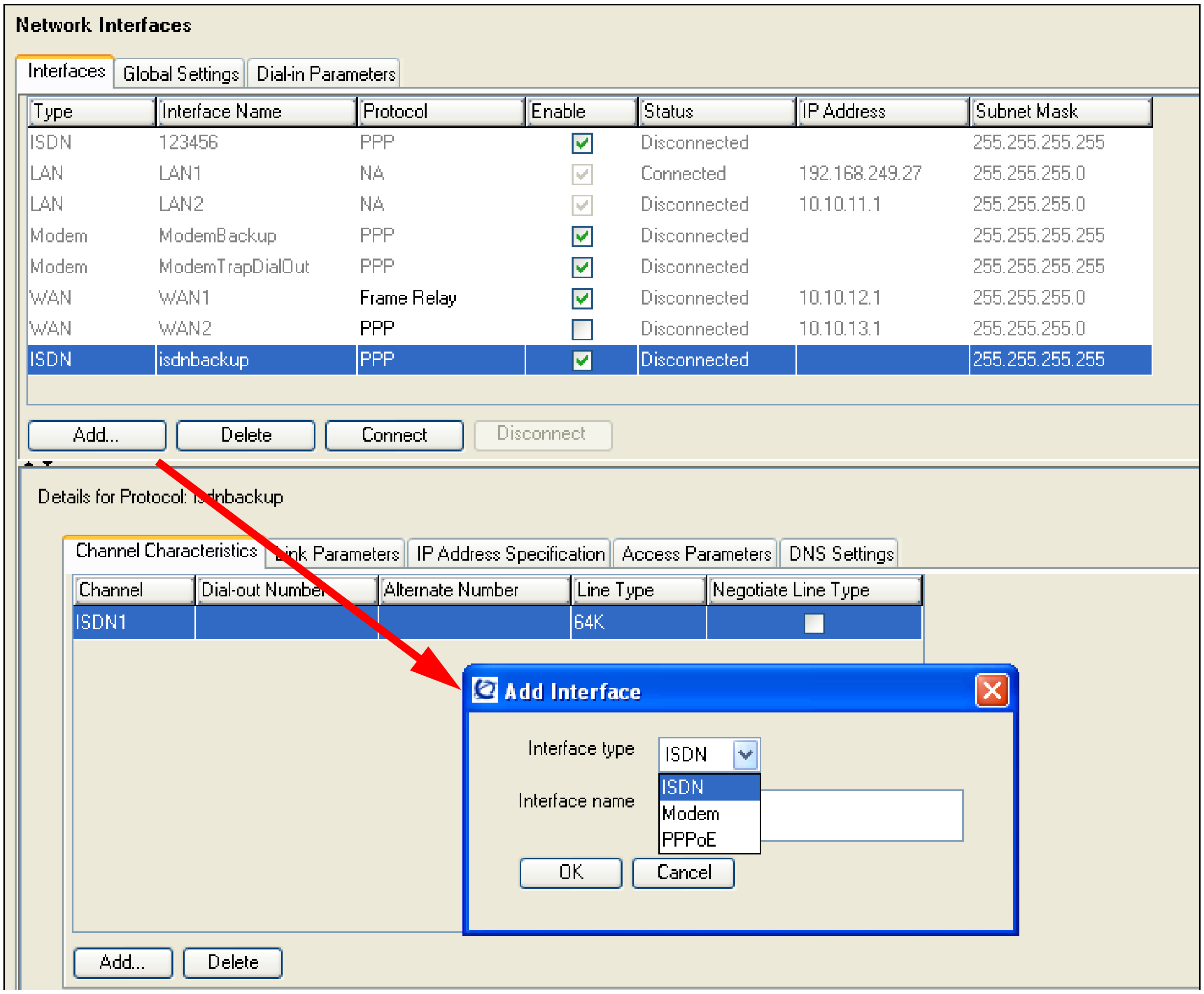
1 Click **Configuration > Resources > Network Interfaces**

2 Click **Add.**

3 The **Add Interface** dialog box appears. See Figure 169.

4 Configure the settings. Refer to the information in Table 131.

Figure 169 Adding a PPPoE Interface



Type	Interface Name	Protocol	Enable	Status	IP Address	Subnet Mask
ISDN	123456	PPP	☑	Disconnected		255.255.255.255
LAN	LAN1	NA	☑	Connected	192.168.249.27	255.255.255.0
LAN	LAN2	NA	☑	Disconnected	10.10.11.1	255.255.255.0
Modem	ModemBackup	PPP	☑	Disconnected		255.255.255.255
Modem	ModemTrapDialOut	PPP	☑	Disconnected		255.255.255.255
WAN	WAN1	Frame Relay	☑	Disconnected	10.10.12.1	255.255.255.0
WAN	WAN2	PPP	□	Disconnected	10.10.13.1	255.255.255.0
ISDN	isdnbackup	PPP	☑	Disconnected		255.255.255.255

Details for Protocol: isdnbackup

Channel	Dial-out Number	Alternate Number	Line Type	Negotiate Line Type
ISDN1			64K	□

Interface type: ISDN
ISDN
Modem
PPPoE

Interface name:

Table 131 Adding a PPPoE Interface

Attribute	Value	Description
Interface type	ISDN Modem PPPoE	Select the type of Interface to add.
Interface name	<text>	User-friendly name of the logical interface being created.

Configuring a PPPoE interface

Configuring a PPPoE interface includes the following settings:

- "Link Parameters" on page 538
- "IP Address Specifications" on page 552

- “DNS Settings” on page 553
- “Access Parameters” on page 554

Link Parameters

To configure the PPPoE Link Parameters

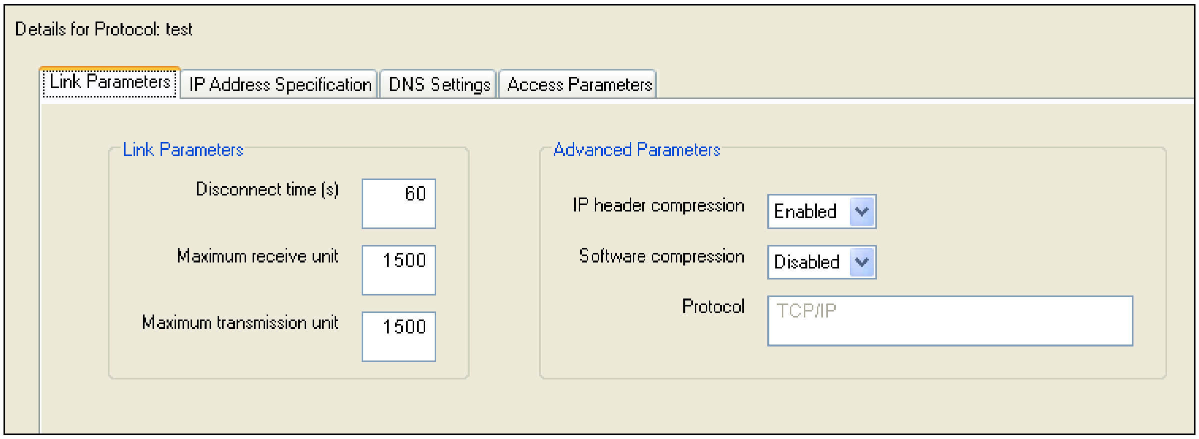
1 Click **Configuration > Resources > Network Interfaces.**

2 Click the **Interfaces** tab.

3 Select the interface you want to modify.
The details panel appears. See Figure 170.

4 Configure the PPPoE Link Parameters. Refer to the information in Table 132.

Figure 170 Link Parameters



Details for Protocol: test

Link Parameters

Disconnect time (s) 60

Maximum receive unit 1500

Maximum transmission unit 1500

Advanced Parameters

IP header compression Enabled

Software compression Disabled

Protocol TCP/IP

Table 132 PPPoE link parameters

Attribute	**Value**	**Description**
Link Parameters		
Disconnect time (s)	Numeric	Shows the interval, in seconds, after which the PPPoE interface disconnects when there is no traffic. The Disconnect Time is set to 0 which means the PPPoE interface will not disconnect.
Maximum receive unit	<128-16384>	The maximum size of the packets that can be received. Default: 1500.
Maximum transmission unit	<128-16384>	The maximum size of the packets that will be sent. Default: 1500.
Advanced Parameters		
IP header compression	Enabled Disabled	Enable or disable IP header compression. The feature must be enable at both ends of the connection. Default: Enabled.
Software Compression	Enabled Disabled	Enable or disable software compression. When enabled, all dial-up connections use Microsoft Point-to-Point Compression (MPPC). Default: Enabled.
Protocol	TCP/IP	Read-only. The protocol that this PPPoE interface uses.

IP Address Specifications

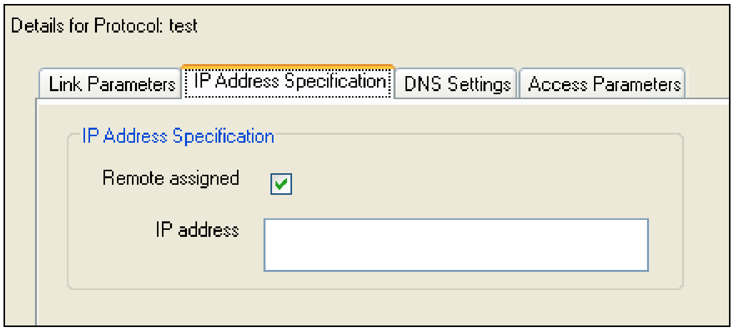
To configure the PPPoE IP Address Specifications

1. Click **Configuration > Resources > Network Interfaces.**
2. Click the **Interfaces** tab.
3. Select the interface you want to modify.
   The details panel appears.

4 Click the **IP Address Specification** tab.
The IP Address Specification panel appears. See Figure 171.

5 Configure the IP Address Specifications. Refer to the information in Table 133.

Figure 171 IP Address Specification



Details for Protocol: test

Remote assigned ☑

IP address

Table 133 IP Address Specification

Attribute	Value	Description
IP Address Specification		
Remote assigned	<check box>	When selected, BCM obtains IP address from the remote end. Default: Enabled.
IP address	<IP Address>	When the Remote assigned check box is cleared, a static IP address needs to be configured in this parameter.

DNS Settings

To configure PPPoE DNS Settings

1 Click **Configuration > Resources > Network Interfaces.**

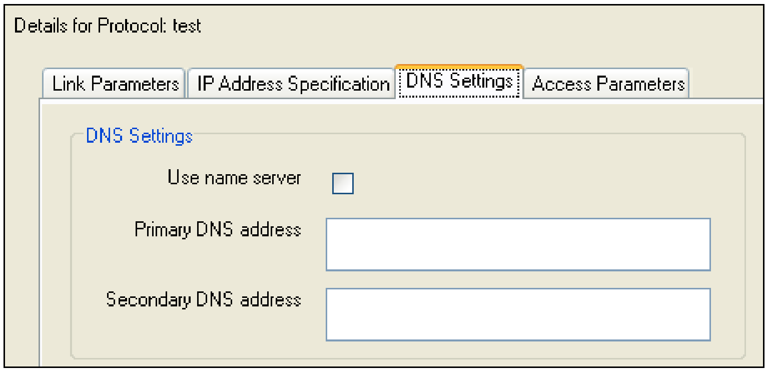
2 Click the **Interfaces** tab.

3 Select the interface you want to modify.
The details panel appears.

4 Click the **DNS Settings** tab.
The DNS Settings panel appears. See Figure 172.

5 Configure the settings. Refer to the information in Table 134.

Figure 172 DNS Settings Panel



Details for Protocol: test

DNS Settings

Use name server □

Primary DNS address

Secondary DNS address

Table 134 DNS Settings

Attribute	Value	Description
DNS Settings		
Use name server	<check box>	Enable DNS Servers configuration for this interface. When the check box is selected, the other two fields become writable. Default: Disabled
Primary DNS address	<IP address>	Enter the IP address of the Primary DNS server that this interface will use.
Secondary DNS address	<IP address>	Enter the IP address of the Secondary DNS server that this interface will use.

Access Parameters

To configure PPPoE Access Parameters

1 Click **Configuration > Resources > Network Interfaces.**

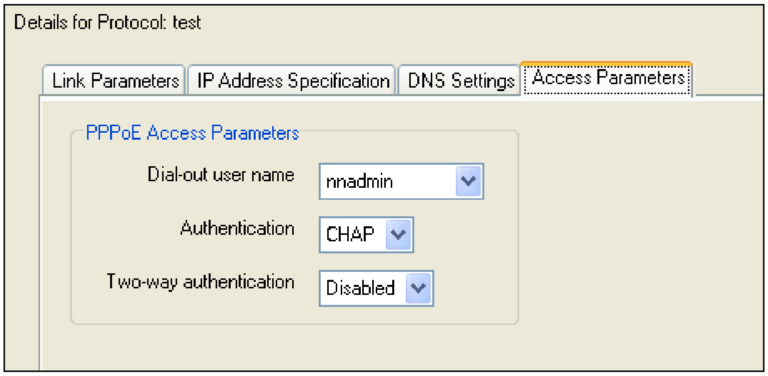
2 Click the **Interfaces** tab.

3 Select the interface you want to modify.
The details panel appears.

4 Click the **Access Parameters** tab.
The **Access Parameters** panel appears. See Figure 173.

5 Configure the Access Parameters. Refer to the information in Table 135.

Figure 173 Access Parameters



Details for Protocol: test

Dial-out user name: nnadmin

Authentication: CHAP

Two-way authentication: Disabled

Table 135 PPPoE access parameters

Attribute	**Value**	**Description**
PPPoE Access Parameters		
Dial-out user name	nnadmin BTRemoteIsdn	The User name used for authenticating to the remote end. Default: nnadmin
Authentication	PAP CHAP	Select the authentication type for the link. The options are **PAP** or **CHAP**. **PAP**: When selected, the PAP is used for authentication. **CHAP**: When selected, CHAP is used for authentication. Default: CHAP.
Two-way authentication	Enabled Disabled	Enable or disable link authentication in both directions. Default: Disabled.

After you configure PPPoE, make sure that your broadband modem is powered up and connected to the LAN1 interface.

Deleting a PPPoE interface

To delete a PPPoE interface

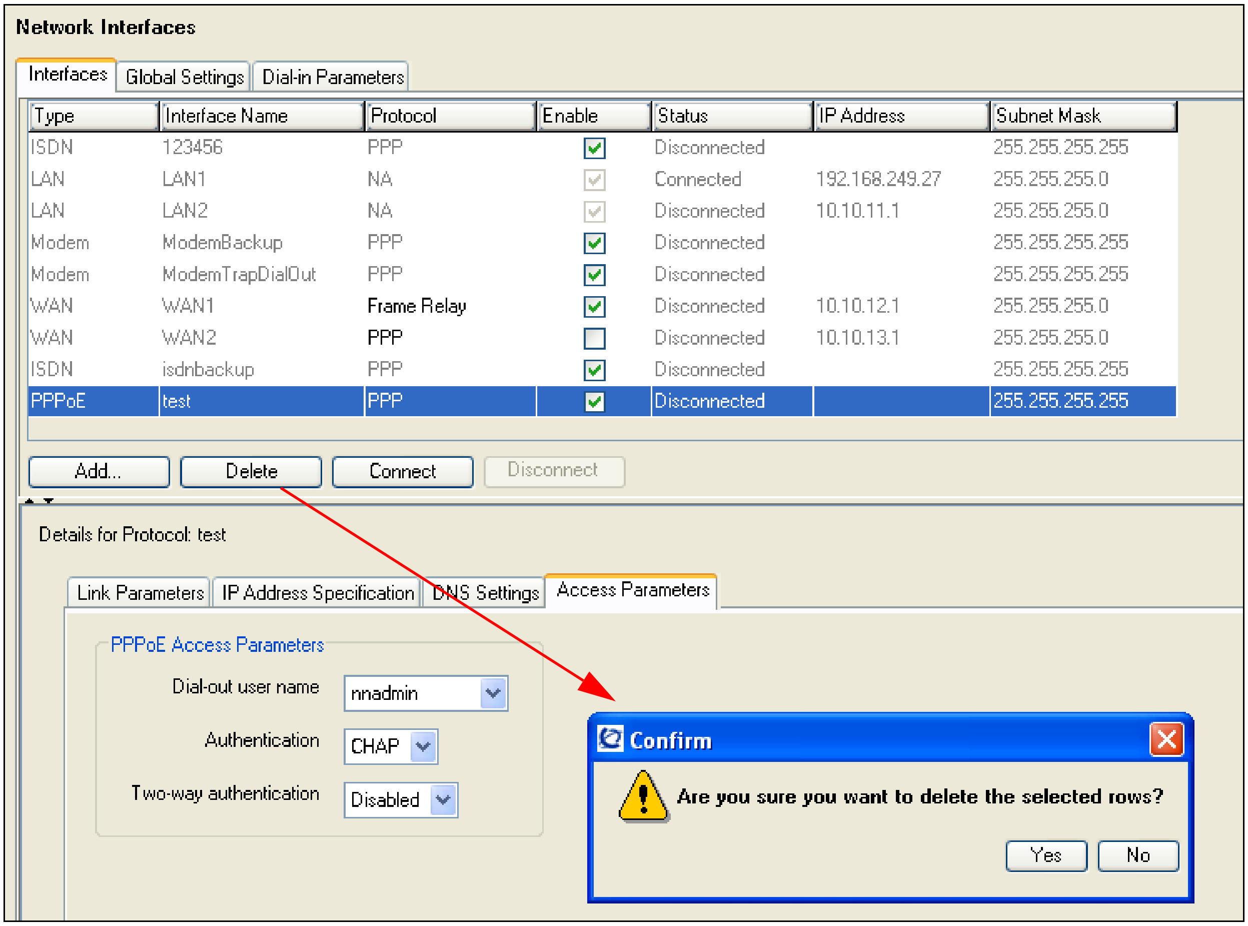
1 Click **Configuration > Resources > Network Interfaces**

2 Select the PPPoE interface you want to delete.
The details panel appears.

3 Click **Delete**.
A confirmation dialog box appears. See Figure 174.

4 Click **Yes**.

Figure 174 Deleting a PPPoE Interface



Type	Interface Name	Protocol	Enable	Status	IP Address	Subnet Mask
ISDN	123456	PPP	☑	Disconnected		255.255.255.255
LAN	LAN1	NA	☑	Connected	192.168.249.27	255.255.255.0
LAN	LAN2	NA	☑	Disconnected	10.10.11.1	255.255.255.0
Modem	ModemBackup	PPP	☑	Disconnected		255.255.255.255
Modem	ModemTrapDialOut	PPP	☑	Disconnected		255.255.255.255
WAN	WAN1	Frame Relay	☑	Disconnected	10.10.12.1	255.255.255.0
WAN	WAN2	PPP	□	Disconnected	10.10.13.1	255.255.255.0
ISDN	isdnbackup	PPP	☑	Disconnected		255.255.255.255
PPPoE	test	PPP	☑	Disconnected		255.255.255.255

Details for Protocol: test

PPPoE Access Parameters

Dial-out user name: nnadmin

Authentication: CHAP

Two-way authentication: Disabled

Are you sure you want to delete the selected rows?

Auto dial-out

Management applications such as SNMP trap dial out, Scheduled Log transfer, Scheduled Backup, and Scheduled CDR records transfer can use automatic dial-out over an ISDN or Modem interface. To configure the automatic data transfer, the administrator must configure a static route, with the auto dial-out field selected, and associate it with the application. When data is sent to the destination address, the network recognizes the address of the application, and triggers the dial-out to establish the connection.The packets are then sent over the link to the destination.

Notes:

- The interface must be enabled to configure static routes.
- The disconnect time for the interface must be greater than 60 seconds. This is configured on the **Link Parameters** tab of the selected interface under **Configuration > Resources > Network Interfaces**.
- Auto dial-out routes cannot be added if the interface is already manually connected, unless the interface is already connected with auto dial-out routes configured.

Creating an auto dial-out interface

To add an auto dial-out Interface

1. Create a Modem or ISDN interface. See "Creating a modem interface" on page 536 or "Creating an ISDN dial out interface" on page 543.
2. Enable the interface under **Configuration > Resources > Network Interfaces**.
3. Set the **Disconnect time (s)** on the **Link Parameters** tab to a value greater than 60 seconds.
4. Add a static route. "To add a static route to the routing table" on page 568.
5. Select the **Auto Dial-out** check box.
6. Associate the route with an application.

To manually disconnect an auto dial-out interface

➡ **Note:** Auto dial-out interfaces are disconnected automatically once data transfer is complete.

1. Click **Configuration > Resources > Network Interfaces**.
2. Select the interface to disconnect.
3. Click **Disconnect**.
   A confirmation dialog box will appear.
4. Click **Yes**.

Guidelines for using remote Dial-in

Consider the following guidelines when using remote dial-in:

- The remote dial-in for administration and the backup WAN link share the same modem. If a remote administration user is connected while the primary link breaks, the automatic backup function does not occur.
- While using the back-up interface, BCM always calls. BCM does not answer an incoming call from a router on the V.92 interface.

Using a dial-up interface as a primary connection

The dial-up interfaces on the BCM are used as a Primary or Secondary interfaces. The BCM does not have default dial-up settings, the Administrator must add them.

The following tasks can be configured to use dial-up as a primary connection:

- SNMP auto trap dial-out

- modem user secure callback
- CDR records retrieval
- backup to a remote destination
- log collection to a remote destination
- software upgrades

The basic steps to set dial-up as the primary connection are:

1. Create or assign an account with remote access privileges.
2. Create a dial-up interface, and enter the username of the account with remote access privileges as the dial-out username.
3. Create a static route for the dial-up interface, or assign a dial-out number, depending on the type of device selected.
4. Tell the application to use the route.

The following example demonstrates how to configure the dial-up interface.

Example: To configure SNMP auto trap dialout

Assign an account remote access privileges

1. Click **Configuration > Administrator Access > Accounts and Privileges >View by Accounts** tab.
2. Click **Add**.
   The **Add Account** dialog box appears. Refer to the *BCM 4.0 Administration Guide* (N0060598) for information on configuring an account.
3. Select the account to which you want to assign remote access privileges.
   The details panel appears.
4. Select the **Group Membership** tab.
5. Click **Add**.
   The **Add Account To Group** dialog box appears.
6. Select **Remote Access** group.
7. Click **OK**.

Create a dial-up interface

1. Click **Configuration > Resources > Network Interfaces > Interfaces**.
2. Click **Add**.
   The **Add Interface** dialog box appears.
3. Select **Modem** from the drop-down menu.
4. Enter a logical name for the interface in the interface name field.
5. Click **OK**.

6 Select the new **Modem** interface.

7 Click the **Link Parameters** tab in the bottom panel.

8 Enter the **Dial-out number** to use for the back-up.

9 Click the **Access Parameters** tab in the bottom panel.

10 Select the account with remote access privileges from the **Dial-out user name** drop-down menu.

11 Select the **Authentication** and **Two-way authentication** values that are appropriate for your configuration. See Table 127 for a description of these fields.

Add the SNMP Trap destination

1 Click **Configuration > Administrator Access > SNMP > SNMP Trap Destinations** tab.

2 Click **Add**.
The **Add Trap Destination** dialog box appears.

3 See the *BCM 4.0 Administration Guide* (N0060598) for information on how to configure the **Add Trap Destination** dialog box.

Note: The Host address must be the IP address of the static route created in this procedure.

4 Click **OK**.

For information on configuring a dial-up resource as a back up for a primary connection refer to "Configuring Net Link Manager" on page 561

Chapter 63
Configuring Net Link Manager

Net Link Manager is a BCM service that provides WAN link failure protection. This is achieved by selecting a primary WAN interface and a back up WAN interface. Only dial up interfaces can be selected as a WAN backup.

Refer to the following for information about enabling/disabling Net Link Manager, as well as how to select WAN links:

- “To enable or disable Net Link Manager” on page 563
- “Selecting a permanent WAN link as the primary WAN connection” on page 563
- “Selecting a dial-up link as the primary WAN connection” on page 565

When Net Link Manager detects a primary WAN link failure, Net Link Manager automatically establishes a backup WAN connection, if one is configured. Net Link Manager monitors the WAN primary link by performing multiple tests. When a predetermined number of tests fails, Net Link Manager establishes the backup connection.

The backup connection uses a V.92 modem or one or more ISDN B-channels or the PPPoE interface. When the backup WAN connection is active, Net Link Manager continues to monitor the status of the primary WAN link connection. When the primary WAN link connection is determined to be available again, Net Link Manager re-establishes the primary WAN link and disconnects/disables the backup connection.

Table 22 Backup Links

		Primary Link						
Interfaces		**Permanent WAN1**	**Permanent WAN2**	**LAN1**	**LAN2**	**PPPoE on LAN1**	**Modem V.90/V.92**	**ISDN**
Backup Link	**Permanent WAN1**	N	N	N	N	N	N	N
	Permanent WAN2	N	N	N	N	N	N	N
	LAN1	N	N	N	N	N	N	N
	LAN2	N	N	N	N	N	N	N
	PPPoE on LAN1	Y	Y	N	Y	N	N	N
	Modem V.90/V.92	Y	Y	Y	Y	N	N	N
	ISDN	Y	Y	Y	Y	N	N	N
Y= Connection is supported. N= Connection is not supported.								

Warning: If dial-up connection is used as the primary WAN connection, no backup link is available.

Note: Net Link Manager manages the default route in BCM. If the primary link fails, Net Link Manager removes the default route from the Primary link and adds it to the backup link. This happens during the switch over from primary to backup link. The default route returns to the primary link after the connection to the primary WAN link is re-established.

To enable or disable Net Link Manager

1 Click **Configuration > Resources > Network Interfaces > Global Settings tab**.
The Net Link Manager Summary panel appears.

2 Select or clear the **Enable Net Link Manager** check box.

Selecting a permanent WAN link as the primary WAN connection

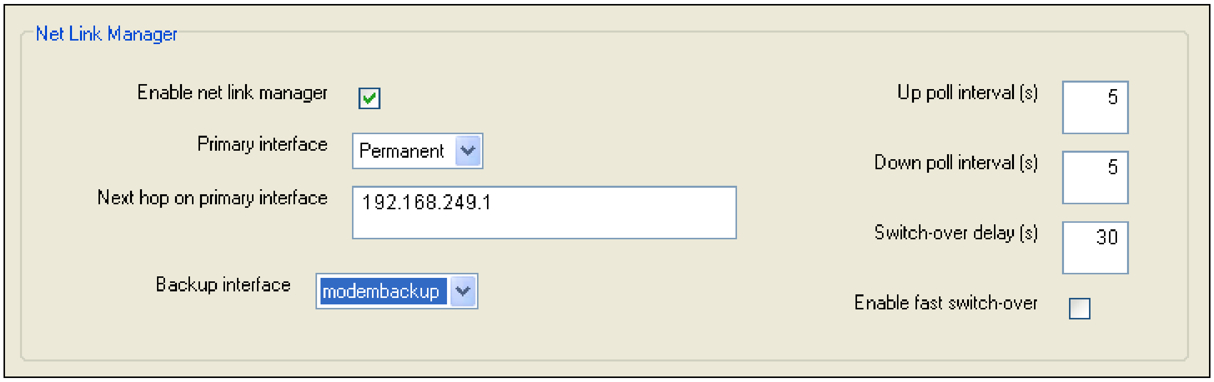
1 Click **Configuration > Resources > Network Interfaces > Global Settings tab**.
The Net Link Manager details panel appears.

2 Select the **Enable net link manager** check box.

3 Select **Permanent** from the **Primary interface** drop-down list.

4 Configure the Permanent WAN Connections Settings. Refer to the information in Table 136.

Figure 175 Permanent WAN connection details panel



Net Link Manager

Enable net link manager ☑

Primary interface: Permanent

Next hop on primary interface: 192.168.249.1

Backup interface: modembackup

Up poll interval (s): 5

Down poll interval (s): 5

Switch-over delay (s): 30

Enable fast switch-over □

Table 136 Permanent WAN connection settings (Sheet 1 of 2)

Attribute	Value	Description
Net Link Manager		
Enable net link Manager	<check box>	Enable or disable Net Link Manager. Default: Enabled
Primary interface	<drop-down list>	Select whether the primary interface is a Permanent or Dialup connection.
Next hop on primary interface	<IP address>	Enter the IP address (in dot format) of the next hop router. This address is used by Net Link Manager to add a default route in BCM. If this address ever becomes unreachable, Net Link Manager dials the backup link and changes the default route. This address also identifies the default IP gateway associated with a default route. This is the IP address of the remote router connected via the primary WAN connection.
Backup interface	<drop-down list>	Select which interface to use for WAN backup. You must configure a backup interface before you can select it. For information about how to create an ISDN backup interface, refer to "Creating an ISDN dial out interface" on page 543.
Up poll interval (s)	<0- 3600000>	Set the polling interval on the Primary WAN Link. The Up poll interval is the interval between successive pings when the next hop on the primary link is available. Default: 5 seconds
Down poll interval (s)	<0- 3600000>	Set the polling interval on the Primary WAN Link, in seconds, when the primary WAN link is down, and the backup (dial-up) WAN link is operational. A short interval provides faster detection of the primary link availability. Default: 5 seconds

Table 136 Permanent WAN connection settings (Sheet 2 of 2)

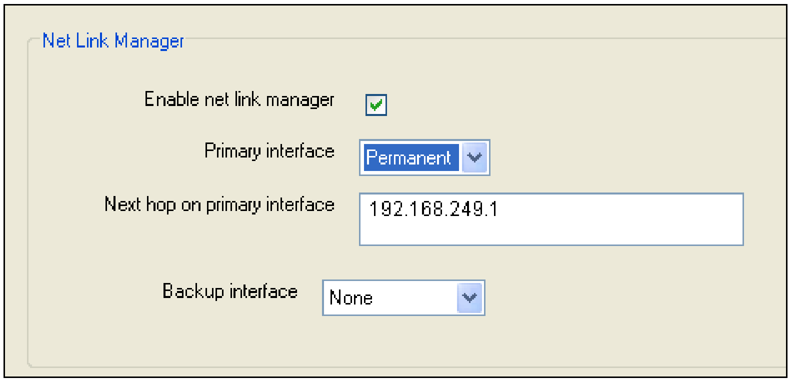
Attribute	Value	Description
Switch-over delay (s)	<0- 3600000>	Define the interval, in seconds, that Net Link Manager waits before switching back to the primary WAN link when it becomes available. This delay is to let the router at the other end of the primary link recognize that the primary link has come up and allows for necessary routing table updates. Default: 30 seconds
Enable fast switch-over	<check box>	Enable or disable the Fast switch-over feature. When Fast switch-over is enabled, BCM uses the Link Status of the WAN interface to quickly detect an interface failure and switch over to the Backup WAN Link, when the Primary WAN Link is down. When Fast switch-over is **Disabled**, BCM pings the remote router if there is no response from the previous attempt. **Note:** If there is no response after five retries, BCM switches over to the Backup WAN Link. When Fast switch-over is **Enabled**, BCM checks the Link Status of the WAN interface when there is no response from the next hop router. If the Link Status is Down, BCM immediately switches over to the Backup WAN link. If the link status is **Up**, BCM attempts to contact the next hop router again. If there is no response after several retries, BCM switches over to the Backup WAN Link. **Note: PPP active/WAN link switch over delay issue:** If you enable Fast Backup Switch Over on a WAN link that is using PPP protocol, change the setting for LCP Keep Alive Interval to 3. This reduces the time it takes for BCM to determine that the WAN link is not functioning to 30 seconds (3 seconds X 10 intervals). If you leave the LCP Keep Alive Interval at its default value, the BCM takes 100 seconds (10 seconds X 10 intervals) to determine that the WAN link is not functioning. **Note**: If BCM receives a response from the next hop router on any attempt, it does not switch over to the Backup WAN Link. **Note**: The Fast switch-over feature does not affect how BCM switches from the Backup WAN Link to the Primary WAN Link. Default: Disabled.

Selecting a dial-up link as the primary WAN connection

The dial-up WAN connection supports PPP only. BCM supports ISDN dial-up and PPPoE dial-up WAN connections.

Setting a dial-up connection as the primary WAN connection means that the BCM default route is to the dial-up connection. If the dial-up WAN connection is configured as dial-on-demand, any traffic across the dial-up WAN connection causes the link to be established. Also, if there is no traffic crossing the connection, the link shuts down automatically after a time out.

Figure 176 Dial-up WAN connection



Net Link Manager

Enable net link manager ☑

Primary interface: Permanent

Next hop on primary interface: 192.168.249.1

Backup interface: None

➡ **Note:** When you configure your primary WAN connection to use a dial-up WAN connection, no backup WAN connection is available.

1. Click **Configuration > Resources > Network Interfaces > Global Settings** tab.
   The Net Link Manager details panel appears.
2. Select the **Enable net link manager** check box.
3. Select **Dialup** from the **Primary interface** drop-down list.

➡ **Note:** Before you can select an ISDN or PPPoE dial-up interface to connect to the network, you must first create the dial-up interface under **Configuration > Resources > Network Interfaces**. For information on creating an ISDN dial-up interface, see "Access Parameters" on page 542. For information on creating a PPPoE dial-up interface, see "Point to Point Protocol over Ethernet (PPPoE)" on page 549.

Chapter 64
Configuring IP Routing

The **IP Routing** service setting allows you to select, add or delete routing protocol on specific interfaces, choose routing protocol options, and add or delete static routes.

Note: If you change the IP address or subnet mask of any interface (LAN or WAN), you must reboot BCM before you configure IP routing.

BCM supports the following IP routing protocols:

- "Static routes" on page 567
- "Routing Information Protocol (RIP)"
- "Open Shortest Path First (OSPF)"

Also refer to the following:

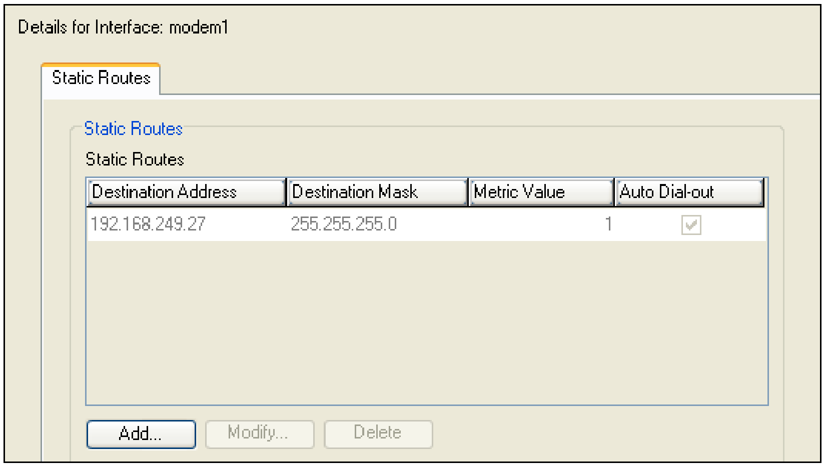
- "Configuring IP Routing global settings" on page 570
- "Configuring IP routing on an interface" on page 573

Static routes

You can add static routes to the BCM routing table. Static routes added to the routing table take precedence over routes added by OSPF. Refer to Table 138 for the list of routing precedence.

Note: The default route is managed by Net Link Manager. For information about Net Link Manager, refer to "Configuring Net Link Manager" on page 561.

Figure 177 Static Routes



Destination Address	Destination Mask	Metric Value	Auto Dial-out
192.168.249.27	255.255.255.0	1	☑

Table 137 IP Static Routes attributes

Attribute	Value	Description
Destination Address	<IP address>	Enter the IP address of the destination network or host.
Destination Mask	<IP address>	Enter the subnet mask corresponding to the destination address.
Metric Value	<1-32767>	Enter the metric value associated with the route. Default: 1.
Auto Dial-out	<check box>	If selected, dialout is triggered when traffic is directed towards the destination address. **Note**: This field is only available for ISDN and modem interfaces.

To add a static route to the routing table

1. Click **Configuration > Data Services > Router**.
2. Click the **Interfaces** tab.
3. Select an interface to add a static route.
4. In the details panel click the **Static Routes** tab.
   The **Static Routes** panel appears.
5. Click **Add**.
   The **Add Static Route** dialog box appears.
6. Configure the static route attributes. Refer to the information in Table 137.
7. Click **OK**.

To modify the static route configuration

> **Note:** Destination Address in static routes cannot be modified. However, the other fields: Destination Mask, Metric Value, and Auto Dial-out can be modified.

1. Click the static route you want to modify in the **Static Routes** table.
2. Click **Modify**.
   The Static Route dialog box appears.
3. Modify the static route attributes.
4. Click **Save**.

To delete a static route

1. Click the static route you want to delete in the Static Route table.
2. Click **Delete**.
   A confirmation message appears.
3. Click **Yes**.

Routing Information Protocol (RIP)

BCM supports RIP, a widely-used protocol for managing routing information in a self-contained network, such as a corporate intranet. RIP measures the shortest path between two points on a network in terms of the number of hops between those points.

BCM router sends RIP routing information updates that list all the other hosts it knows about, to its nearest neighbor host every 30 seconds. The neighbor host sends the information to its next neighbor, until all the hosts in the network know the routing paths, a state known as network convergence. RIP uses a hop count to determine network distance. Each router in the network uses the routing table information to determine the next host for the packet, until it reaches the destination.

BCM does not support on demand routing table update. It supports periodic routing table update.

> **Note:** IP subnet aggregation is not supported.

For information on how to select RIP as your routing protocol, see “Configuring RIP parameters on a network interface” on page 573.

Open Shortest Path First (OSPF)

Open Shortest Path First protocol bases its path descriptions on “link states” that take into account additional network information. OSPF also lets the user assign cost metrics to a given host router so that some paths are given preference. For information on how to select OSPF as your routing protocol, see “Configuring OSPF Parameters on a network interface” on page 577.

The implementation of OSPF on BCM is designed to operate as an edge router in an OSPF intranet, or as a backup router in a small network. Do not configure BCM for multiple OSPF areas.

Note: BCM is an edge router and will not act as a router spanning RIP and OSPF routing networks (RIP or OSPF redistribution).

IP routing protocol precedence

The following table shows the BCM IP routing protocols and the precedence order when conflicting or redundant routes occur.

Table 138 IP routing protocol precedence

Precedence	IP Routing Protocols
1	Static Routes added by IPSec
2	Static Routes added by Net Link Manager
3	OSPF v2 routes
4	Static routes added by user from Element Manager
5	RIP v1 and v2

Configuring IP Routing global settings

The following describes how to configure global settings for IP Routing.

It also includes information about:

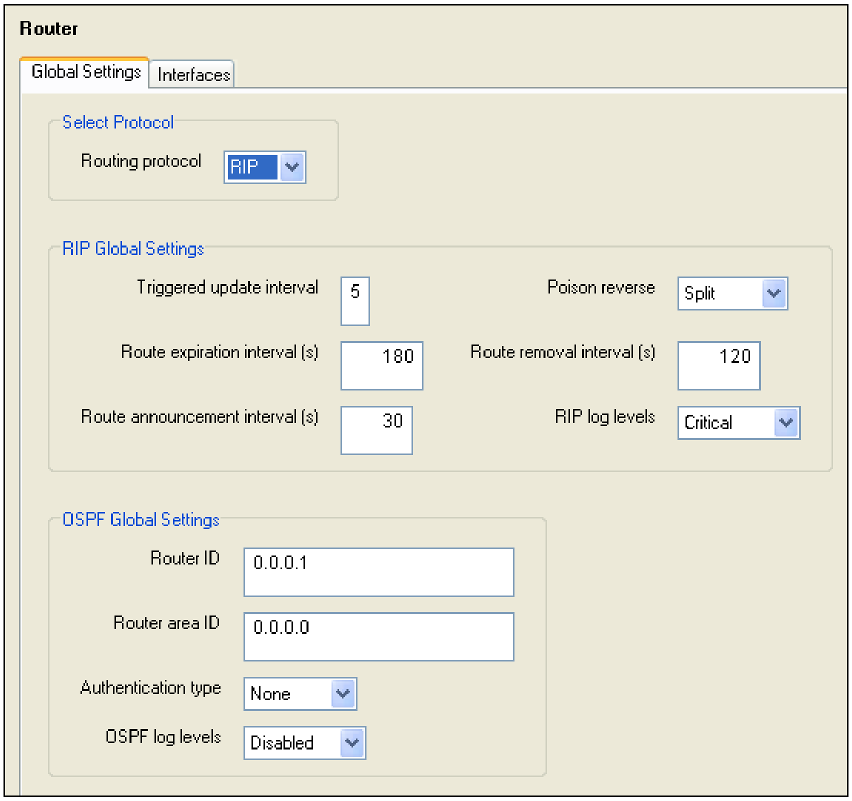
- “Configuring RIP parameters on a network interface”
- “Configuring OSPF Parameters on a network interface” on page 577

To configure global settings for IP Routing

1 Click **Configuration > Data Services > Router**.
The Router panel appears. See Figure 178.

2 Configure the Global Settings. Refer to the information in Table 139.

Figure 178 IP Routing Global Settings



Select Protocol

Routing protocol: RIP

RIP Global Settings

Triggered update interval: 5

Poison reverse: Split

Route expiration interval (s): 180

Route removal interval (s): 120

Route announcement interval (s): 30

RIP log levels: Critical

OSPF Global Settings

Router ID: 0.0.0.1

Router area ID: 0.0.0.0

Authentication type: None

OSPF log levels: Disabled

Table 139 IP Routing settings (Sheet 1 of 3)

Attribute	Value	Description
Select Protocol		
Routing Protocol	None RIP OSPF	Select the routing protocol to use. Default: None
RIP Global Settings		
Triggered update interval	<1-5>	The minimum interval, in seconds, at which a router must send routing tables update if the metric for a given route changes. If the router detects a change in the routing information, the router sends an update message at the specified interval. Default: 5 secs.
Route expiration interval (s)	<15– 259200>	Set the period of time a route in the routing table must be updated to remain a valid route. Default:180 secs.
Route announcement interval (s)	<5-86400>	Set the time interval (in seconds) between routing update messages sent by the router. Default: 30 secs

Table 139 IP Routing settings (Sheet 2 of 3)

Attribute	Value	Description
Poison reverse	Split Poisoned Actual	This feature is used to enable or disable options designed to avoid routing problems such as loops or metric values exceeding the maximum of 15 hop counts. This command is used to avoid including routes in updates sent to the same gateway from which they were learned. Using the split horizon command omits routes learned from one neighbor, in updates sent to that neighbor. Using the poisoned parameter with this command includes such routes in updates, but sets their metrics to infinity. Thus, advertising that these routes are not reachable. Split (split horizon): A routing table update process designed to avoid sending the same routing information back to the originator. Poisoned: A routing table update process designed to advertise unreachable routes as having metric value of 16 regardless of incoming routing update information. Actual: Disable the techniques designed to avoid routing problems like routing loops and slow reconvergence. Default: Split
Route removal interval (s)	<15– 259200>	Define the period of time an invalid route remains in the routing table before the routing manager removes it from the routing table. Default: 120 secs.
RIP log levels	All Critical Warning Information Disabled	Enable the recording of events in the Operational Log. **All,** for logging all information in the Element Manager. **Critical** for logging critical errors and warnings in the Element Manager. **Warning**, for logging warnings in the Element Manager. **Information** for logging Information events in the Element Manager. **Disabled,** to disable event logs. Default: Disabled
OSPF Global Settings		
Router ID	<IP address>	Specify the IP address that uniquely identifies the router on the network.
Router area ID	<IP address>	Define the area ID for OSPF interfaces. Default: 0.0.0.0

Table 139 IP Routing settings (Sheet 3 of 3)

Attribute	Value	Description
Authentication type	None Password	Specify the type of OSPF authentication used. If the Authentication Type component is set to “Password” directly connected OSPF interfaces between 2 routers will establish an OSPF adjacency only if they use the same password. **Note**: If password is selected as the authentication type, interfaces can still have blank/empty passwords set on both sides. Default: None
OSPF log levels	All Critical Warning Information Disabled	Enable the recording of events in the Operational logs. **All,** for logging all information in the Element Manager. **Critical** for logging critical errors and warnings in the Element Manager. **Warning**, for logging warnings in the Element Manager. **Information** for logging Information events in the Element Manager. **Disabled,** to disable event logs. Default: Disabled

Configuring IP routing on an interface

After you configure the IP Routing global settings, you must configure each available network interface to use the routing protocol of your choice or static routes.

Note: You must use the same routing protocol on all interfaces. For example, you can not configure your LAN1 interface to use RIP and your WAN1 interface to use OSPF.

The following provides instructions on how to configure interfaces for IP routing and how to create static routes. The available interfaces appear under the **IP Routing** heading. Follow the same instructions to configure all interfaces. For information on how to create static routes, see “Static routes” on page 567.

Refer to the following topics:

- “Configuring RIP parameters on a network interface”
- “Configuring OSPF Parameters on a network interface” on page 577
- “OSPF NBMA Neighbors” on page 580
- “Static routes” on page 567

Configuring RIP parameters on a network interface

To configure RIP parameters on a network interface

1 Click **Configuration** > **Data Services** > **Router** > **Global Settings tab.**

2 Select **RIP** from the Routing protocol drop-down list.

3 Click the **Interfaces** tab.
The available interfaces for IP routing are listed under the **Interfaces** panel.

4 Click the interface you want to configure.
The Summary panel appears. The **Enable Routing Protocol** check boxes indicate which interfaces are enabled.

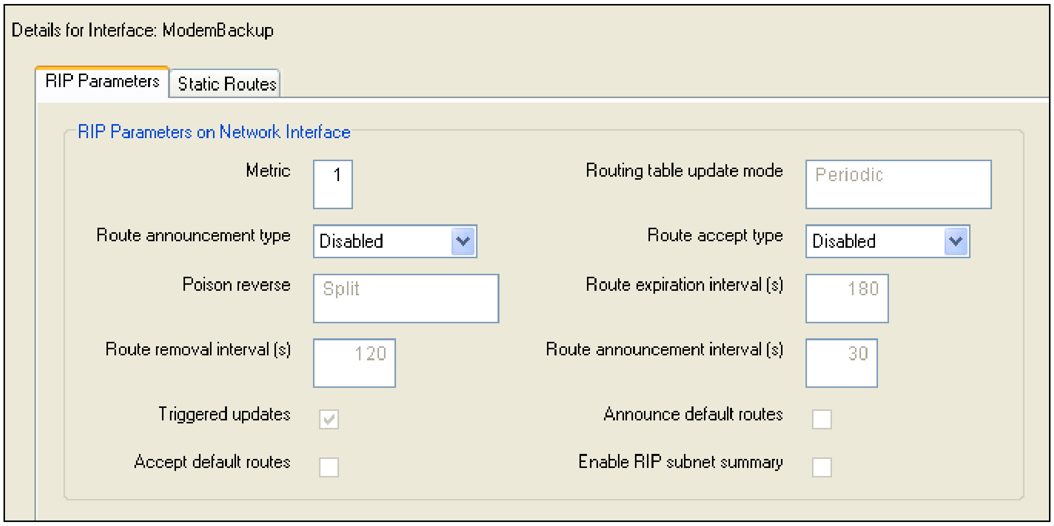
Tips
If you are changing the routing protocol from OSPF to RIP, you must first clear the **Enable Routing Protocol** beside each available interface before you can select **RIP**.

Note: The RIP Parameters window does not appear unless you choose RIP as your routing protocol in the Global Settings tab.

5 In the **Interfaces** list, select the check box beside the interface to enable.
The RIP Parameters tab appears. See Figure 179.

6 Configure the RIP Parameters. Refer to the information in Table 140.

Figure 179 RIP parameters



Details for Interface: ModemBackup

RIP Parameters | Static Routes

RIP Parameters on Network Interface

Metric	1	Routing table update mode	Periodic
Route announcement type	Disabled	Route accept type	Disabled
Poison reverse	Split	Route expiration interval (s)	180
Route removal interval (s)	120	Route announcement interval (s)	30
Triggered updates	☑	Announce default routes	□
Accept default routes	□	Enable RIP subnet summary	□

Table 140 IP RIP Parameters (Sheet 1 of 3)

Attribute	**Attribute**	**Description**
RIP Parameters on Network Interfaces		
Metric	<1-16>	Assign a cumulative value (in terms of hop count or associated cost) to route updates received through this interface. The Routing Manager adds to the metric value of each route learned through this interface, the metric value of this interface and uses the result to select the best route to a destination. Because RIP protocol can handle up to 15 hop counts before reaching destination, a value of **16** corresponds to “counting to infinity”. Default: 1.
Route announcement type	Disabled RIP1 RIP1 Compatible RIP2	Set the type of routing table update announcements the BCM router sends to other routers. The possible values are: **Disabled**: disables sending RIP routing update. If you choose **Disabled**, you must configure the other routers in the subnet to use static routes to access the Business Communications Manager base unit. **RIP1**: sends only announcements of RIP v1 type in broadcast mode. **RIP1 Compatible**: sends RIP v1 and RIP v2 packets in broadcast mode. Use this for a network environment that uses RIP v1 and RIP v2. **RIP2**: sends RIP v2 packets in multicast mode only. Use this type of announcement only if all other routers connected to the Business Communications Manager base unit support RIP v2. Default: RIP1 compatible.

Table 140 IP RIP Parameters (Sheet 2 of 3)

Attribute	Attribute	Description
Poison reverse	Actual Split Poisoned	Enable or disable options designed to avoid routing problems such as loops or metric values exceeding the maximum of 15 hop counts. The following options are available: **Actual**: A routing table update process where a routing table update going out repeats the information sent by the originator. The system tries to solve this state known as a loop involving two routers by sending more routing updates. **Split** (split horizon): A routing table update process designed to avoid sending the same routing information back to the originator. **Poisoned**: A routing table update process designed to advertise unreachable routes as having metric value of 16 regardless of incoming routing update information. **Note:** This field can be modified on the **Global Settings** tab. Default: **Split**.
Route removal interval (s)	<read-only>	Read-only. This field reflects the Route removal interval value set under **Data Services > Router Global Setting > RIP Global settings**.
Announce default route	<read-only>	Read-only. Enable or disable the announcement of default routes in outgoing route announcements. Default: Disabled. **Note:** This feature is not supported in this release of BCM.
Enable RIP subnet summary	<read-only>	Read-only. **Note:** This feature is not supported in this release of BCM.
Routing table update mode	<read-only>	Read-only field. The value is set to **Periodic**. The router sends its table to other routers running RIP at regular intervals.
Route accept type	Disabled RIP1 RIP1 Compatible RIP2	Set the type of routing table update announcements the BCM router accepts from other routers. The possible values are: **Disabled**: If you choose **Disabled**, you must create static routes in the Business Communications Manager base unit to access other networks connected to this interface. This method is preferable if you want to keep the routing table small in the Business Communications Manager base unit. **RIP1**: accepts only announcements of RIP 1 type. **RIP1 Compatible**: accepts announcements of RIP 1 and RIP 2 types. **RIP2**: accepts announcements of RIP 2 type only. Default: **RIP1 Compatible**.
Route expiration interval (s)	<read-only>	Read-only. This field will reflect the Route expiration interval value set under **Configuration > Data Services > Router > Global Settings > RIP Global Settings**.
Route announcement interval (s)	<read-only>	Read-only. This field will reflect the Route announcement interval value set under **Configuration > Data Services > Router > Global Settings > RIP Global Settings**.

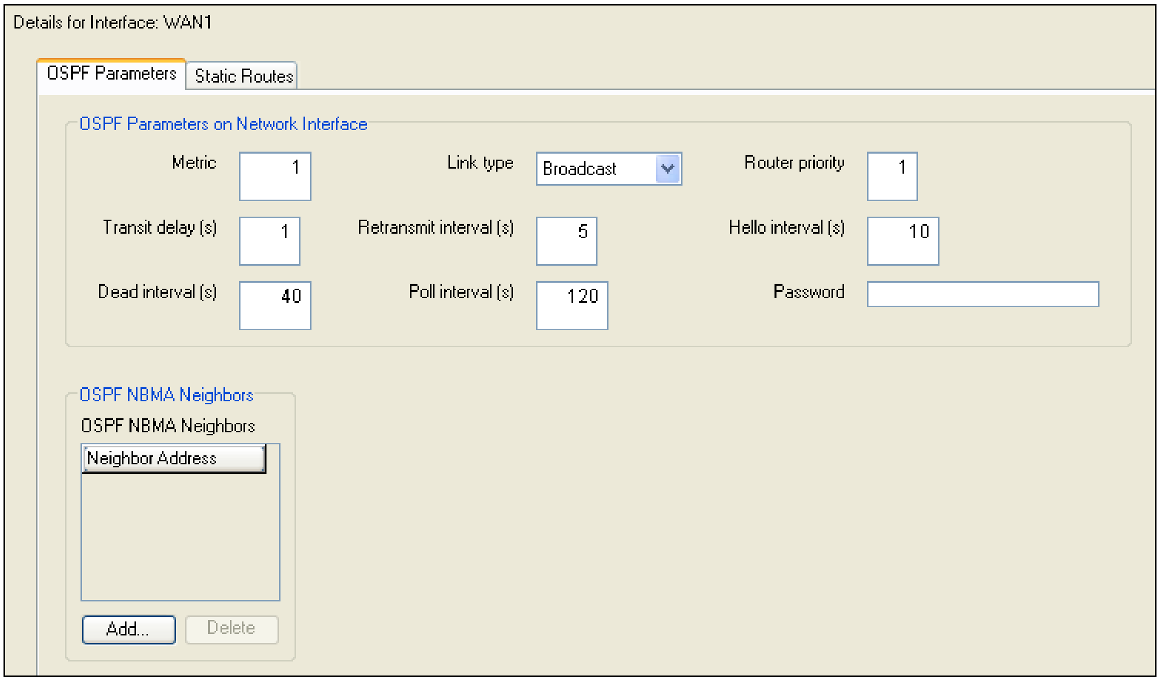
Table 140 IP RIP Parameters (Sheet 3 of 3)

Attribute	Attribute	Description
Triggered updates	<read-only>	Read-only. Immediate route update announcements are sent whenever a metric or other information changes in the routing table entries. The system gathers new routing information for the period of time defined in the **Triggered Update Interval** from the RIP Summary window (see "Configuring IP Routing global settings" on page 570). Triggered updates results in more frequent, smaller RIP routing table updates. Default: **Enabled**.
Accept default route	<read-only>	Read-only. **Note:** This feature is not supported in this release of BCM.

Configuring OSPF Parameters on a network interface

1. Click **Configuration > Data Services > Router**.
2. Select **OSPF** from the **Routing Protocol** drop-down list on the **Global Settings** tab.
3. Select the **Interfaces** tab.
4. Click **LAN/WAN**.
   The details panel appears.
5. Click the **OSPF Parameters** tab.
   The OSPF Parameters panel appears. See Figure 180.
6. Configure the OSPF parameters. Refer to the information in Table 141.

Figure 180 OSPF parameters



Details for Interface: WAN1

Metric: 1

Link type: Broadcast

Router priority: 1

Transit delay (s): 1

Retransmit interval (s): 5

Hello interval (s): 10

Dead interval (s): 40

Poll interval (s): 120

Password:

Tips
If you are changing the routing protocol from RIP to OSPF, you must first clear the **Enable Routing Protocol** check boxes before you can select OSPF.

Note: The **OSPF Parameters** tab does not appear unless you choose OSPF as your routing protocol.

Table 141 IP OSPF Parameters

Attribute	Attribute	Description
OSPF Parameters on Network Interface		
Metric	<1-32767>	Assign the link cost for this interface that advertised in the router's link state advertisement for this interface. The Metric is an indication of the cost of the route. If multiple routes exist on a network ID, the Metric is used to decide which route is taken. The route with the lowest Metric is the preferred route. If you enter a high number for the Metric, this interface will not be used as much as an interface with a lower Metric. Default:1.
Transit delay (s)	<1-3600 >	Set the estimated round-trip transit delay in the network connected to the interface. Default: 1.
Dead Interval	<1-32767>	Set the maximum number of seconds the router waits to receive the next hello before considering the adjacent router as non operational. Default: 40.
Link type	Broadcast Point -To-Point NBMA	Select the type of interface that describes your network configuration. The possible values are: **Broadcast**: A broadcast network supports multiple routers and can send a broadcast/multicast message to all routers. **Point-To-Point**: A point-to-point network joins a single pair of OSPF routers. **NBMA**: A Non-Broadcast-Multi-Access (NBMA) network supports multiple routers and cannot broadcast/multicast to all routers. Default: Broadcast.
Retransmit interval (s)	<1-3600>	Set the number of seconds the router waits before retransmitting after a time-out occurs. Default: 5.
Poll interval	<1-32767>	Define the period of time the router must keep sending hello packets to an adjacent router that is considered non operational. Default:120.
Router priority	<0-255>	Assign a priority to the BCM router. A value of 0 indicates that the BCM system cannot become the designated router. Default: 1.
Hello interval (s)	<1-32767>	Define how frequently the router must send "hello packets" on an interface. Default:10.
Password	<alphanumeric>	Define an authentication password, if you selected **Password** as the authentication type in the **Authentication Type** box on the OSPF Global Parameters window. There is no default value provided because the **Authentication Type** is set to **None** by default. **Note**: The password you choose must conform to the password policy used for your BCM system.

OSPF NBMA Neighbors

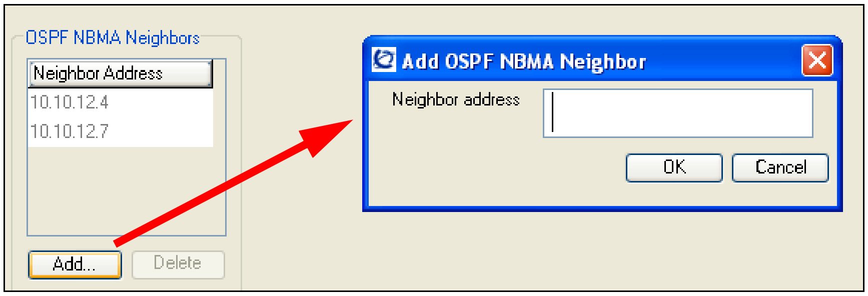
Frame Relay on BCM is a Non Broadcast Multiple Access (NBMA) network. NBMA is a network that can connect two or more routers, but has no hardware broadcast capability. For OSPF to function properly on an NBMA network, you must configure the router running OSPF to send unicast control packets to the OSPF neighbor routers. The OSPF NBMA Neighbors panel allows you to enter IP addresses of the NBMA Neighbors.

Note: The OSPF NBMA Neighbors is available only for WAN interfaces.

1 Click **Configuration > Data Services > Router**.

2 Click the **WAN** interface you want configure.
The OSPF NBMA Neighbors field appears under the OSPF parameters tab.

Figure 181 Add OSPF NBMA Neighbors


OSPF NBMA Neighbors

Neighbor Address
10.10.12.4
10.10.12.7

Table 142 IP OSPF NBMA Neighbor parameters

Attribute	Attribute	Description
Neighbor Address	<IP address>	Specify the IP address of the neighboring router. **Note:** This address must be on the same subnet.

To add OSPF NBMA Neighbors

1 Click **Add**.
The **Add OSPF NBMA Neighbor** dialog box appears. See Figure 181.

2 Enter the Neighbor address.

3 Click **OK**.

To delete OSPF NBMA Neighbors

1 Click the OSPF NBMA Neighbor you want to delete.

2 On the **Configuration** menu, click **Delete OSPF Neighbor**.
A confirmation dialog box appears.

3 Click **Yes**.

Restarting the router

When you change dynamic routing protocols, the router is automatically restarted. There will be a slight service interruption while the static routes forward the traffic. When the dynamic routing protocol returns (either OSPF or RIP), it will take precedence over static routes and normal traffic forwarding will resume

➡ **Note:** This procedure will affect any service that requires access across the LAN or WAN, including IP telephone service.

Chapter 65
Configuring DHCP

BCM provides DHCP (Dynamic Host Configuration Protocol) service to branch office clients. DHCP allows a network administrator to supervise and distribute IP addresses from a central location. This service dynamically assigns IP addresses to branch office computers or IP telephones, so you do not need to manually assign an IP address. It also automatically assigns a new IP address if a device connects to a different place in the network.

Refer to the following topics for more information:

- “Configuring the DHCP Mode” on page 584
- “Configuring a DHCP Server” on page 584
- “LAN settings for DHCP Server” on page 587
- “Remote Scope” on page 594
- “Configuring a DHCP Relay Agent” on page 599
- “LAN settings for DHCP Relay Agent” on page 601
- “IP Terminal DHCP Options” on page 603

DHCP configuration overview

To configure BCM as your DHCP server, you must create a scope of IP addresses for each LAN interface and then allocate a block of IP addresses to that scope. If you already have a DHCP server then you need to set up BCM as a relay agent to that server.

Caution: Check with your network administrator before enabling DHCP. Enabling DHCP on a network that already has a DHCP server can cause problems on the network.

Tip
Because BCM retrieves default DHCP parameters from the LAN interface parameters, you must configure a LAN interface before you configure the DHCP server for that interface. For information on configuring a LAN interface, see “Configuring the LAN resources” on page 487.

You must define one DHCP scope for each LAN interface. For DHCP service, there are global attributes that affect all scopes and there are attributes that are specific for each scope.

Tip
Use the BCM DHCP default configuration unless your network does not allow it.

If you must modify the DHCP default configuration on BCM, make sure configuration settings are consistent throughout the network and take the following into consideration:

- If a change in the DHCP configuration resulted in a change in the IP addresses of a scope, perform one of the following actions to ensure good system operation:

— Execute *ipconfig/release and ipconfig/renew* on each of the workstations. For Windows 95 and Windows 98, use the *winipcfg*.

— For clients that do not support *ipconfig* and *winipcfg* (for example, IP telephones), a reboot is necessary to renew their IP addresses.

- If you made a change in the DNS server configuration or DNS name field, repeat the actions stated in the previous step to ensure proper connectivity with the network.
- Always schedule a down time when making changes to the BCM DHCP server configuration to minimize impact on your network users.

Configuring the DHCP Mode

You can configure BCM as your DHCP Server or as a DHCP Relay Agent by setting the DHCP Mode.

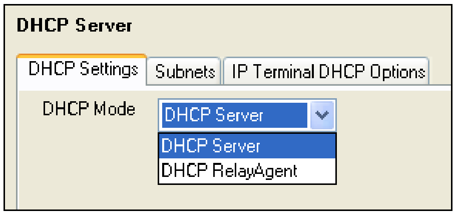
- Choose DHCP Server mode if you want BCM to supply the IP addresses to the devices on your network.
- Choose DHCP RelayAgent if you have a central DHCP Server on your corporate network and you want BCM to pass the DHCP traffic to and from the devices on the LAN.

To set the DHCP Mode

1 Click **Configuration > Data Services > DHCP Server**.
The DHCP Settings panel appears.

2 Select **DHCP Server** or **DHCP RelayAgent** from the DHCP Mode drop-down list.
The default is DHCP Server.

Figure 182 DHCP Mode - DHCP Server



Configuring a DHCP Server

If you chose DHCP Server as the DHCP mode, configure the DHCP Server settings in the General Settings Panel under the Subnets tab.

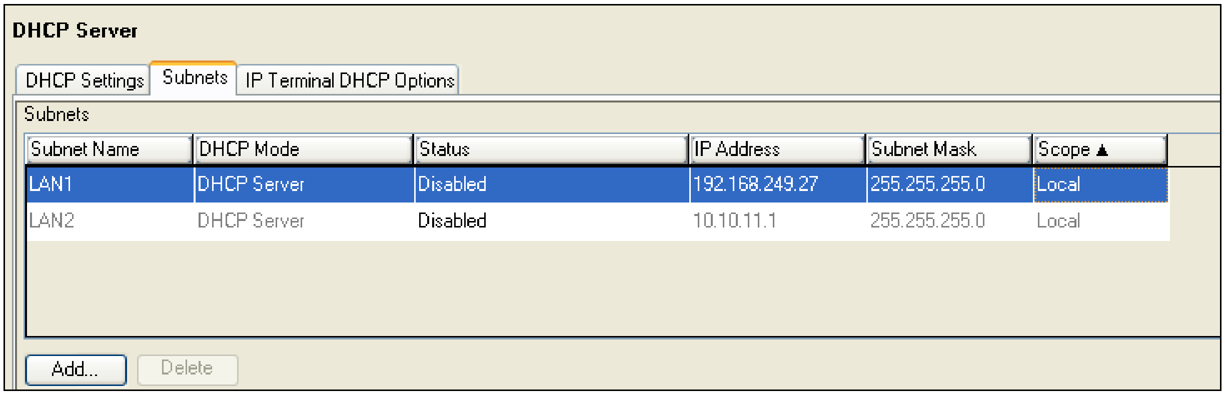
1 Click **Configuration > Data Services > DHCP Server > Subnets tab**.
The Subnets panel appears. See Figure 183.

2 Configure the Subnet Settings. Refer to the information in Table 183.

3 Select the Subnet to configure from the subnets table.
A details panel appears.

4 Configure the General Settings attributes. Refer to the information in Table 144.

Figure 183 Subnets



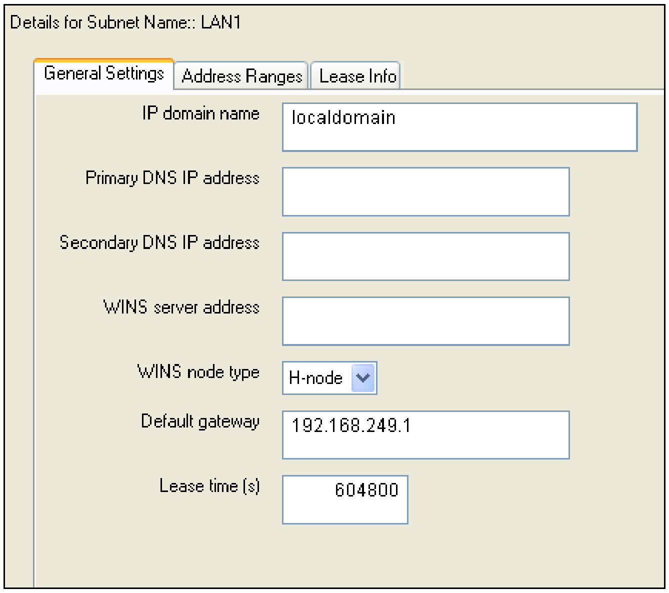
DHCP Server

Subnet Name	DHCP Mode	Status	IP Address	Subnet Mask	Scope ▲
LAN1	DHCP Server	Disabled	192.168.249.27	255.255.255.0	Local
LAN2	DHCP Server	Disabled	10.10.11.1	255.255.255.0	Local

Table 143 Subnets

Attribute	Definition
Subnet Name	Name given to identify this Subnet
DHCP Mode	DHCP Mode chosen during server configuration
Status	Select whether the DHCP server is disabled or enabled. If the server is enabled select one of the following options: • Enabled - Automatic • Enabled - IP Phones Only • Enabled - All Devices
IP Address	IP address of the subnet
Subnet Mask	Subnet mask of the subnet
Scope	Specifies the scope of the subnet if it is Local or Remote.

Figure 184 General Settings



Details for Subnet Name:: LAN1

IP domain name: localdomain

Primary DNS IP address:

Secondary DNS IP address:

WINS server address:

WINS node type: H-node

Default gateway: 192.168.249.1

Lease time (s): 604800

Table 144 DHCP General Settings (Sheet 1 of 2)

Attribute	Definition
IP domain name	This setting defaults to the value entered in the **Domain** box of the DNS Summary page (see "Configuring DNS" on page 679) because all the DHCP clients of BCM are in the same DNS domain as the BCM. BCM runs only a DNS cache and does not introduce another DNS zone. The domain name is passed to the client when BCM responds to a client's DHCP requests. **Note:** Use caution if you change this.
Primary DNS IP address	Specify the IP address of the primary DNS to be used by DHCP clients.
Secondary DNS IP address	Specify the IP address of the secondary DNS to be used by DHCP clients.
WINS server address	Specify the address of the Windows Internet Server, which resolves IP addresses on a DHCP network.
WINS node type	Specify a client's WINS node type. BCM automatically sets this value to H-Node on all DHCP clients of BCM. This setting configures the DHCP client PCs to use P-node name resolution before resorting to B-node name resolution. This is efficient when a WINS server is configured for the network. BCM also includes a WINS server. Other options available are: • B-node that uses a broadcast mechanism for NetBIOS name resolution. • P-node that uses a point-to-point mechanism involving a WINS Server for NetBIOS name resolution. • M-node that first uses a broadcast and then a point-to-point mechanism for NetBIOS name resolution. Default: H-node
Default gateway	Specify the IP address of the default next-hop router. BCM automatically assigns the value for this parameter. Generally, this is the IP address of the BCM next-hop router for public LAN interface. For private LAN interface and remote subnets it will be the IP Address of the corresponding subnet.

Table 144 DHCP General Settings (Sheet 2 of 2)

Attribute	Definition
Lease time (s)	Specify the time, before a DHCP lease expires and the device must request a new IP address. Default: 604800 seconds (1 week).

➡ **Note:** When you change the published IP address of BCM, you must reboot the IP telephones.

⚠ **Warning:** Whenever you make changes to the default gateway, the DHCP server may become unavailable to clients for a brief period of time. When making changes, consider doing so at a time that will minimize the effect on users.

LAN settings for DHCP Server

Refer to the following:

- "Configuring Address ranges for a Local Scope" on page 588
- "Configuring Reserved addresses for a Local Scope" on page 591

If you configured the DHCP mode for DHCP Server, then configure the LAN scope attributes as follows. If the DHCP mode is DHCP RelayAgent refer to "LAN settings for DHCP Relay Agent" on page 601.

➡ **Note:** If your BCM system has multiple LAN interfaces, you can see multiple DHCP scopes under DHCP. They are named LAN1 and LAN2. This section describes configuring DHCP for LAN1. Follow the same instructions to configure the parameters for LAN2.

➡ **Note:** When DNS is disabled in BCM, the DNS Server box must be set to the IP address of a remote DNS server.

Configuring Address ranges for a Local Scope

⚠ **Warning:** Whenever you make changes to the address range, the DHCP server may become unavailable to clients for a brief period of time. When making changes, consider doing so at a time that will minimize the effect on users.

Address ranges allow you to specify the valid IP addresses for the DHCP clients.

➡ **Note:** You must add at least one Address range to use DHCP server.

By default, the DHCP server on the BCM will have to configure a range of IP addresses to supply the IP Sets. It defaults to use the top 60% of a subnet. For example, if an external DHCP server supplies the following IP address to the BCM: 177.218.21.45 with a subnet of 255.255.255.0, then the DHCP server will configure itself to reserve the following range 177.218.21.100-177.218.21.254. If the IP address of the subnet falls inside the factory default range or the maximum hosts that can be supported is less than 254, then the maximum range which can fit and excludes the IP address of the subnet will be chosen.

Example:

IP address 177.218.21.145 with a subnet of 255.255.255.0

- The following range is reserved 177.218.21.1 - 177.218.21.144

IP address 177.218.21.145 with a subnet 255.255.255.192

- The following range is reserved 177.218.21.146 - 177.281.21.190

This default can be checked and changed using Element Manager.

To add an address range

1. Click **Configuration > Data Services > DHCP Server**.
2. Click the **Subnets** tab and click the **LAN1** heading.
   The details panel appears.
3. Click the **Address Ranges** tab.
   The Address Range panel appears.
4. Click **Add**.
   The Add Included Address Range dialog box appears. See Figure 185.
5. Configure the Address Range attributes. Refer to the information in Table 145.
6. Click **OK** to save the address range.

Figure 185 Add Included Address Range

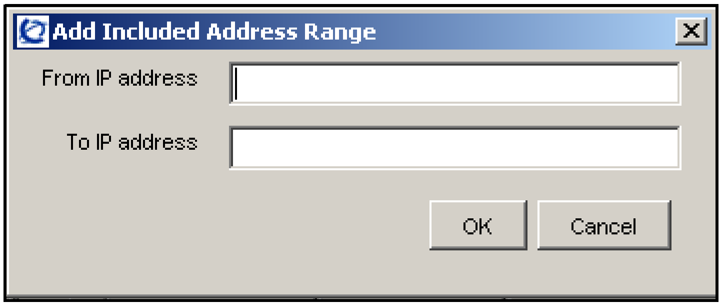


Table 145 Add Included Address Range

Attribute	Definition
From IP Address	Specify the first IP address in the Address Range.
To IP Address	Specify the last IP address in the Address Range. Make sure the start address and end address are in the same subnet.

Note: You cannot add excluded addresses in an address range. Instead, you can use multiple address ranges:

- Create one address range for the IP addresses below the excluded addresses.
- Create a second address range for the IP addresses above the excluded addresses.

  For example, to create an address range from 10.10.10.10 to 10.10.10.49, but excluding addresses from 10.10.10.20 to 10.10.10.29, create one address range from 10.10.10.10 to 10.10.10.19 and one address range from 10.10.10.30 to 10.10.10.49.

To modify an address range

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Subnets** tab and click the **LAN1** heading.
The details panel appears.

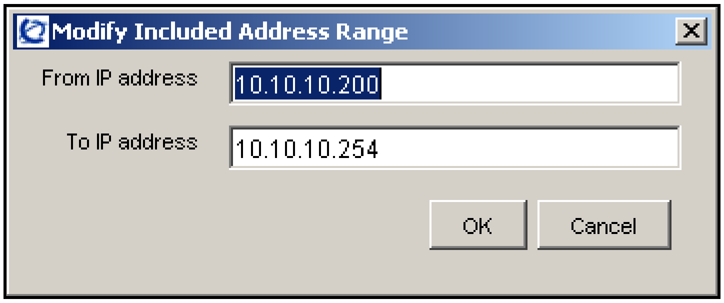
3 Click the **Address Range** tab.
The Address Range panel appears.

4 Click **Modify**.
The Modify Included Address Range dialog box appears. See Figure 186.

5 Modify the Address Range settings. Refer to the information in Table 146.

6 Click **OK**.

Figure 186 Modify Included Address Range



From IP address 10.10.10.200

To IP address 10.10.10.254

Table 146 Modify Included Address Range

Attribute	Definition
From IP Address	Specify the first IP address in the Address Range.
To IP Address	Specify the last IP address in the Address Range. Make sure the start address and end address are in the same subnet.

To delete an address range

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Subnets** tab and click the **LAN1** heading.
The details panel appears.

3 Select the **Address Range** to delete.

4 Click **Delete**.
A dialog box appears asking you to confirm the deletion.

5 Click **Yes**.

Configuring Reserved addresses for a Local Scope

Reserved addresses allow you to assign IP addresses to specific DHCP clients.

You can use Reserved Addresses to assign IP addresses to devices that require a static IP address.

To add a reserved address

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Subnets** tab and click the **LAN1** heading.
The details panel appears.

3 Click the **Address Range** tab.
The Address Range panel appears.

4 Click **Add**.
The Add Reserved Address dialog box appears. See Figure 187.

5 Configure the Reserved Address settings. Refer to the information in Table 147.

6 Click **OK**.

Figure 187 Add Reserved Addresses

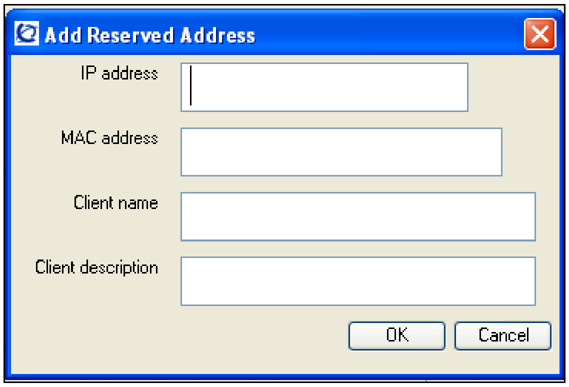


Table 147 Add Reserved Addresses

Attribute	Definition
IP Address	Specify the IP Address that is reserved for this DHCP client.
MAC Address	Specify the MAC address for the DHCP client this IP address is assigned to. The permitted value is 6 bytes in hexadecimal format.
Client name	Specify the name of the DHCP client.
Client description	Specify the description that will help to identify the DHCP client to which this IP address is assigned.

Modifying a reserved address

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Subnets** tab and click the **LAN1** heading.
The details panel appears.

3 Click the **Address Range** tab.
The Address Range panel appears.

4 Click a reserved address in the Reserved Address table.

5 Click **Modify**.
The Modify Reserved Address panel appears.

6 Modify the Reserved Address settings.

7 Click **OK**.

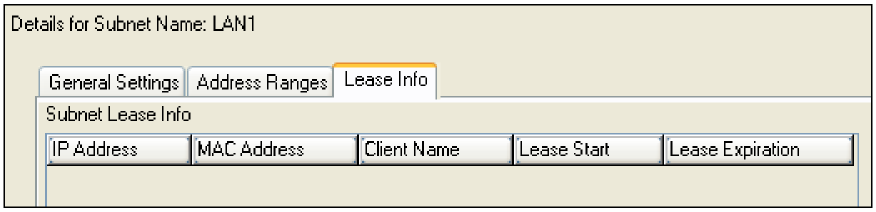
Deleting a reserved address

1. Click **Configuration > Data Services > DHCP Server**.
2. Click the **Subnets** tab and click the **LAN1** heading.
   The details panel appears.
3. Click the **Address Range** tab.
   The Address Range panel appears.
4. Click a reserved address in the Reserved Address table.
5. Click **Delete**.
   A message prompts you to confirm the deletion.
6. Click **Yes**.

Viewing the Lease Information

1. Click **Configuration > Data Services > DHCP Server**.
2. Click the **Subnets** tab and click the **LAN1** heading.
   The details panel appears.
3. Click the **Lease Info** tab.
   The Subnet **Lease Info** panel appears. See Figure 188.
4. Table 148 describes the Lease Information that you can view.

Figure 188 Lease Info



Details for Subnet Name: LAN1

Subnet Lease Info

IP Address	MAC Address	Client Name	Lease Start	Lease Expiration

Table 148 Lease Information

Setting	Definition
IP Address	Shows the IP Address that is reserved for this DHCP client.
MAC Address	Shows the MAC address for the DHCP client this IP address is assigned to.
Client Name	Shows the name of the DHCP client.
Lease Start	Shows the date when this IP address is leased for the DHCP client.
Lease Expiration	Shows the time when this IP address is no longer leased for the DHCP client.

Remote Scope

A remote scope is a remote network (not LAN1 or LAN2) that uses the DHCP Server to get IP addresses through a DHCP relay agent.

Refer to the following topics:

- "Remote Scope" on page 594
- "Modifying Remote Scope general settings" on page 595
- "Configuring Address ranges for a Remote Scope" on page 595
- "Configuring Remote Scope Reserved Addresses" on page 597
- "Remote Scope Lease Information" on page 598
- "To delete a Remote Scope reserved address" on page 599

Remote Scope

To add a Remote Scope

1 Click **Configuration > Data Services > DHCP Server**.

2 Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click **Add**.
The Add Remote Scope Subnet Information panel appears. See Figure 149.

5 Configure the remote scope settings. Refer to the information in Table 149.

6 Click **OK**.

Figure 189 Add Remote Scope Subnet Information

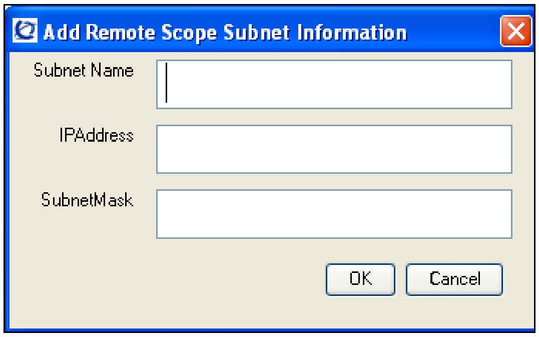


Table 149 Remote Scope settings

Attribute	Definition
Subnet name	Specify the name of the remote scope.
IP Address	Specify the IP address of the remote scope.
Subnet Mask	Specify the subnet mask for the remote scope.

Modifying Remote Scope general settings

1. Click **Configuration > Data Services > DHCP Server**.
2. Select **DHCP Server**.
3. Click the **Subnets** tab.
4. Click the remote scope you want to modify.
   The Scope Specific details panel appears.
5. Configure the remote scope-specific settings. Refer to the information in Table 144.

Configuring Address ranges for a Remote Scope

> **Warning:** Whenever you make changes to the address range, the DHCP server may become unavailable to clients for a brief period of time. When making changes, consider doing so at a time that will minimize the effect on users.

The DHCP server on the BCM will have to configure a range of IP addresses to supply the IP Sets. There is no default address range for Remote Subnets.

Address ranges allow you to specify the valid IP addresses for these remote DHCP clients.

To add a remote scope address range

1. Click **Configuration > Data Services > DHCP Server**.
2. Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click the remote scope you want to modify.
The Scope Specific details panel appears.

5 Click the **Address Ranges** tab.
The Address Range panel appears.

6 On the **Configuration** menu, click **Add**.
The Address Range dialog box appears.

7 Configure the Address Range attributes. Refer to the information in Table 150.

8 Click **OK**.

Table 150 Remote Scope Address Range attributes

Attribute	Definition
From IP Address	Specify a the first IP address in the Address Range.
To IP Address	Specify a the last IP address in the Address Range. Make sure the start address and end address are in the same subnet.

Note: You cannot add exclude addresses in an address range. Instead, you can use multiple address ranges:

- Create one address range for the IP addresses below the excluded addresses.
- Create a second address range for the IP addresses above the excluded addresses.
For example, to create an address range from 10.10.10.10 to 10.10.10.49, but excluding addresses from 10.10.10.20 to 10.10.10.29, create one address range from 10.10.10.10 to 10.10.10.19 and one address range from 10.10.10.30 to 10.10.10.49

To modify a remote scope address range

1 Click **Configuration > Data Services > DHCP Server**.

2 Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click the remote scope you want to modify.
The Scope Specific Options details panel appears.

5 Click the **Address Ranges** tab.
The Address Range panel appears.

6 Click an address in the Address Range table.

7 Click **Modify.**
The Address Range dialog box appears.

8 Modify the Address Range settings.

9 Click **Save**.

To delete a remote scope address range

1 Click the **Services** key and click the **DHCP** key.

2 Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click the remote scope you want to modify.
The Scope Specific Options details panel appears.

5 Click the **Address Ranges** tab.
The Address Range details panel appears.

6 Click an address range in the Address Range table.

7 Click **Delete**.
A dialog box appears asking you to confirm the deletion.

8 Click **Yes**.

Configuring Remote Scope Reserved Addresses

Reserved addresses allow you to assign IP addresses to specific DHCP clients.

To add a remote scope reserved address

1 Click **Configuration > Data Services > DHCP Server**.

2 Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click the remote scope you want to modify.
The Scope Specific Options details panel appears.

5 Click the **Address Ranges** tab.
The details panel appears.

6 Click **Add** under the **Reserved Address** table.
The Reserved Address dialog box appears.

7 Configure the Reserved Address settings. Refer to the information in Table 151.

8 Click **Save**.

Table 151 Remote Scope Reserved Addresses (Sheet 1 of 2)

Attribute	Definition
IP Address	Specify the IP Address that is reserved for this DHCP client.
MAC Address	Specify the MAC address for the DHCP client this IP address is assigned to. The permitted value is 6 bytes in hexadecimal format.

Table 151 Remote Scope Reserved Addresses (Sheet 2 of 2)

Attribute	Definition
Client Name	Specify the name of the DHCP client.
Client Description	Specify the description that will help to identify the DHCP client this IP address is assigned to.

To delete a remote scope reserved address

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Remote Scope** key.

3 Click the remote scope you want to modify.
The Scope Specific Options panel appears.

4 Click the **Reserved Address** tab.
The Reserved Address panel appears.

5 Click a reserved address in the Reserved Address table.

6 On the **Configuration** menu, click **Delete Reserved Address**.
A message prompts you to confirm the deletion.

7 Click **Yes**.

Remote Scope Lease Information

To view the Lease information

1 Click **Configuration > Data Services > DHCP Server**.

2 Select **DHCP Server**.

3 Click the **Subnets** tab.

4 Click the remote scope you want to modify.
The Scope Specific Options details panel appears.

5 Click the **Lease Info** tab.
The **Subnet Lease Info** panel appears. See Figure 188.

6 Table 152 describes the Lease Information you can view for the Subnet.

Table 152 Lease Information for a Remote Scope Subnet

Attribute	Definition
IP Address	Shows the IP Address that is reserved for this DHCP client.
MAC Address	Shows the MAC address for the DHCP client this IP address is assigned to. The permitted value is 6 bytes in hexadecimal format.
Client Name	Shows the name of the DHCP client.
Client Description	Shows the description that will help to identify the DHCP client to which this IP address is assigned.

Table 152 Lease Information for a Remote Scope Subnet

Attribute	Definition
Lease Start	Shows the date when this IP address is no longer reserved for the DHCP client.
Lease Expiration	Shows the time when this IP address is no longer reserved for the DHCP client.

To delete a Remote Scope reserved address

1. Click **Configuration > Data Services > DHCP Server**.
2. Select **DHCP Server**.
3. Click the **Subnets** tab.
4. Click the remote scope you want to delete.
   The Remote Scope Specific Options details panel appears.
5. Click **Delete**.
   A message prompts you to confirm the deletion.
6. Click **Yes**.

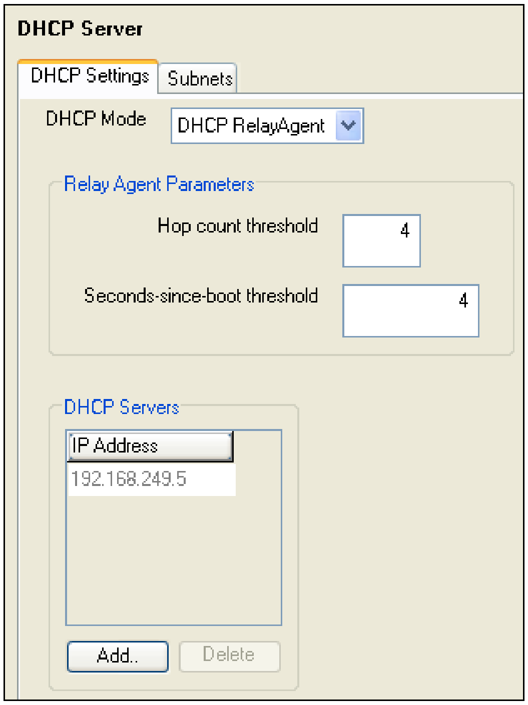
Configuring a DHCP Relay Agent

If you chose DHCPRelayAgent as the mode, configure the DHCP Relay Agent settings in the Global Options and Server List panels with the following process.

1. Click **Configuration > Data Services > DHCP Server**.
   The DHCP Settings panel appears.
2. Select **DHCP RelayAgent** from the **DHCP Mode** drop-down list.
3. Configure the settings. Refer to the information in Table 153.

Note: You can specify a number of servers. The routing component searches the list for the server on the same subnet as the interface and forwards the DHCP packet.

Figure 190 DHCP Mode - DHCP Relay Agent



DHCP Mode: DHCP RelayAgent

Hop count threshold: 4

Seconds-since-boot threshold: 4

IP Address: 192.168.249.5

Table 153 DHCP Relay Agent

Setting	Definition
Relay Agent Parameters	
Hop count threshold	Specify the maximum number of hops. After this number of hops, DHCP requests are discarded. Range: 0 to 16. Default: 4.
Seconds-since-boot threshold	Specify the minimum number of seconds since the last boot of BCM, before BCM forwards DHCP requests. Range:1 to 3600. Default: 4.
DHCP Servers	
IP Address	Enter the IP Address of the DHCP server.

DHCP Server list

To add a server to the DHCP Server list

1. Click **Configuration > Data Services > DHCP Server**.
   The DHCP Settings panel appears.
2. Click **Add** and enter the IP Address of the DHCP Server.
3. Click **OK**.

To delete a server from the DHCP Server List

1 Click **Configuration > Data Services > DHCP Server**.
The DHCP Settings panel appears.

2 Click the IP Address of the DHCP Server you wish to delete.

3 Click **Delete**.
A dialog box appears asking you to confirm the deletion.

4 Click **Yes**.

LAN settings for DHCP Relay Agent

If you configured the DHCP mode as DHCPRelayAgent (refer to "Configuring the DHCP Mode" on page 584), then configure the LAN scope attributes as follows. If the mode is DHCPServer refer to "LAN settings for DHCP Server" on page 587.

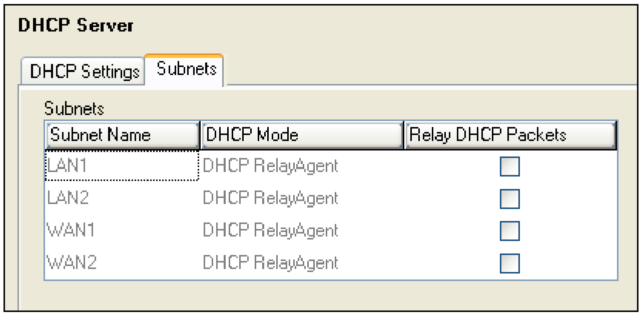
1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **Subnets** tab.
The Subnets panel appears.

➡ **Note:** If your BCM system has multiple LAN interfaces, you can see multiple DHCP scopes under DHCP. They are named LAN1 and LAN2. This section describes configuring the DHCP scope for LAN1. Follow the same instructions to configure any of the parameters under the scope for LAN2.

3 Configure the Subnets. Refer to the information in Table 154.

Figure 191 DHCP Relay Agent Subnets


DHCP Server

Subnets

Subnet Name	DHCP Mode	Relay DHCP Packets
LAN1	DHCP RelayAgent	□
LAN2	DHCP RelayAgent	□
WAN1	DHCP RelayAgent	□
WAN2	DHCP RelayAgent	□

Table 154 Relay Agent Subnets

Attribute	Definition
Subnet Name	Identifies the Subnet.
DHCP Mode	Identifies the DHCP Mode - Server or Relay Agent.
Relay DHCP Packets	Enable or disable the relay of DHCP packets on this interface. Default: Disabled.

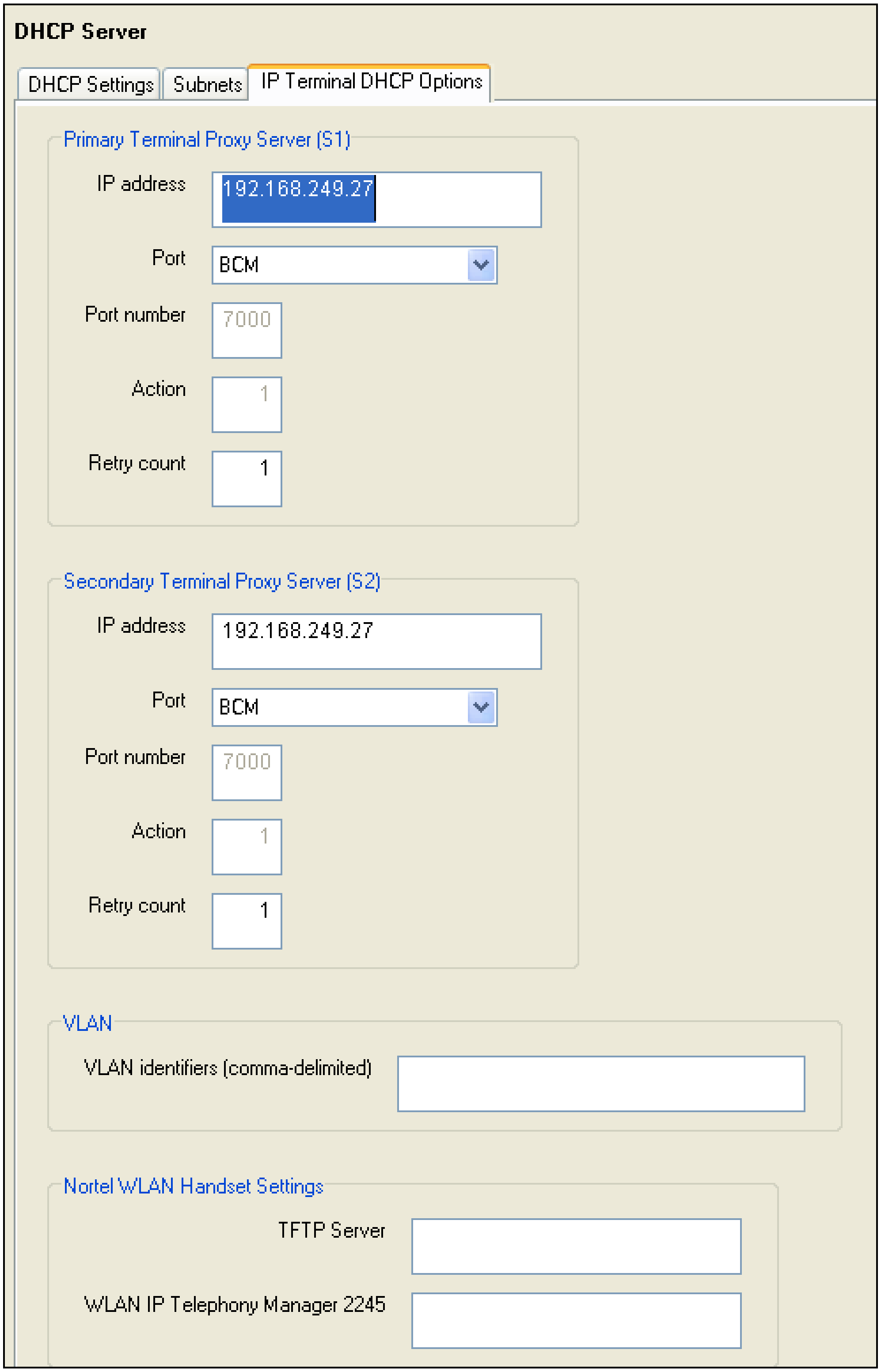
Configuring IP Terminal DHCP Options

1 Click **Configuration > Data Services > DHCP Server**.

2 Click the **IP Terminal DHCP Options** tab. See Figure 192.

3 Configure the settings. Refer to the information in Table 155.

Figure 192 IP Terminal DHCP Options



Primary Terminal Proxy Server (S1)

IP address	192.168.249.27
Port	BCM
Port number	7000
Action	1
Retry count	1

Secondary Terminal Proxy Server (S2)

IP address	192.168.249.27
Port	BCM
Port number	7000
Action	1
Retry count	1

VLAN

VLAN identifiers (comma-delimited)

Nortel WLAN Handset Settings

TFTP Server

WLAN IP Telephony Manager 2245

Table 155 IP Terminal DHCP Options (Sheet 1 of 2)

Attribute	**Definition**
Primary Terminal Proxy Server (S1)	
IP Address	Shows the IP Address that is reserved for this Proxy server.
Port	Select the appropriate port: • BCM • SRG • Meridian1/Succession 1000 • Centrex/SL-100 • Other
Port Number	Displays the port number for the selected Port
Action	The initial action code for the IP telephone.
Retry Count	The retry count for attempts to connect to the TPS (S1).
Secondary Terminal Proxy Server (S2)	
IP Address	Shows the IP Address that is reserved for this Proxy server.
Port	Select the appropriate port: BCM • SRG • Meridian1/Succession 1000 • Centrex/SL-100 • Other
Port Number	Displays the port number for the selected Port
Action	The initial action code for the IP telephone.
Retry Count	The retry count for attempts to connect to the TPS (S2).
VLAN	
VLAN identifiers (comma-delimited)	Specify the Virtual LAN (VLAN) ID numbers that are given to the IP telephones. If you want DHCP to automatically assign VLAN IDs to the IP telephones, enter the VLAN IDs in the following format: **VLAN-A:id1,id2,...,idn.** where: • VLAN-A: — is an identifier that tells the IP telephone that this message is a VLAN discovery message. • id1,id2,...idn — are the VLAN ID numbers that DHCP can assign to the IP telephones. You can have up to 10 VLAN ID numbers listed. The VLAN ID numbers must be a number from 0 to 4095. For example, if you wanted to use VLAN IDs 1100, 1200, 1300 and 1400, you would enter the following string in this box: **VLAN-A:1100,1200,1300,1400.** If you do not want DHCP to automatically assign VLAN IDs to the IP telephones, enter **VLAN-A:none.** in this text box. **Note1**: The NORTEL IP Terminal VLAN Id string must be terminated with a period (.). **Note2**: If you do not know the VLAN ID, contact your network administrator. **Note3**: For information about how to setup a VLAN, refer to the user documentation that came with your VLAN compatible switch.
Nortel WLAN Handset Settings	

Table 155 IP Terminal DHCP Options (Sheet 2 of 2)

Attribute	**Definition**
TFTP Server	Specify the IP address of the TFTP server that is used by WLAN IP telephones. If your system does not have WLAN IP telephones, leave this box empty. Enter the IP address in a valid dot format.
WLAN IP Telephony Manager 2245	Specify the address of the SVP sever that is used by WLAN IP phones.

Chapter 66
Configuring NAT (Network Address Translation)

BCM provides security and firewall features to protect your private data resources from outsiders.

The following links provide information about the different types of NAT:

- "Enabling and disabling NAT" on page 608
- "Configuring an Interface with NAT" on page 608

The Network Address Translation feature is a network security feature. NAT translates the IP addresses used within your private network to different IP addresses known to Internet users outside your private network. NAT helps ensure network security because each outgoing or incoming request must go through a translation process that also provides the opportunity to qualify or authenticate the request or match it to a previous request. NAT also translates port numbers.

NAT is defined by creating a set of rules and then defining the order in which these rules are evaluated.

BCM supports both static and dynamic NAT for a number of packet types and protocols:

NAT Support for:	Type
Packets (static and dynamic)	IP, TCP, UDP, TCP/UDP
Protocols	ALL, FTP, Telnet, SMTP, SNMP-TRAP, DNS, TFTP, Gopher, Finger, H.323, SIP, HTTP, HTTPS, POP3, NNTP, SUNRPC, SUNNFS, UNISTIM, CUSTOM

Static NAT

Static NAT is the one-to-one mapping of an IP address on your private network to an IP address from outside your network. Inbound rules must have external IP addresses mapped to specific internal IP addresses.

Dynamic NAT

Dynamic NAT is the mapping between a private network and the outside network, of one address to a pool of addresses, a pool of addresses to one address or a pool of addresses to another pool of addresses. The mappings are made in a translation table and remain there until the table is cleared or until an entry times out.

Note: When using an inbound translation, be sure that all private addresses belong to the existing systems.

NAT and IP Policy filters

When you use NAT and IP Policy filters, there are two interactions you need to be aware of:

- On inbound traffic, the NAT rules are applied before the IP Firewall Filter rules.
- On outbound traffic, the IP Firewall Filter rules are applied before the NAT rules.

Managing BCM

You cannot manage a BCM system through another BCM system when it is on the Private side of a NAT enabled interface.

Enabling and disabling NAT

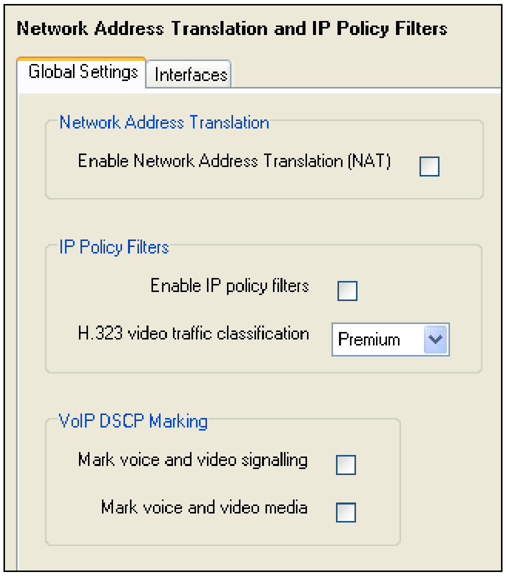
To enable or disable NAT

1 Click **Configuration > Data Services** > **NAT and Filters**.
The Network Address Translation and IP Policy Filters panel appears.

2 Select the **Enable Network Address Translation (NAT)** check box. See Figure 193.
See

3 Clear the check box to disable NAT.

Figure 193 Enable Network Address Translation



Network Address Translation and IP Policy Filters

Network Address Translation

Enable Network Address Translation (NAT) □

IP Policy Filters

Enable IP policy filters □

H.323 video traffic classification: Premium

VoIP DSCP Marking

Mark voice and video signalling □

Mark voice and video media □

Configuring an Interface with NAT

The following describes how to configure an interface with NAT. It also includes information about:

- "To select Default rules" on page 609
- "Adding a Rule to an interface" on page 610

- “Modifying a Rule to an Interface” on page 613
- “Deleting a Rule to an Interface” on page 614

To select Default rules

Note: Default rules are read-only.

1 Click **Configuration > Data Services > NAT and Filters > Interfaces tab**.

2 Click the interface you want to configure.

Note: Rules can be configured in several ways, using default rules, setting up individual rules or a combination of the two.

3 Double-click a **Default NAT Rules** entry to display the drop-down list.
The Rule Order panel appears. See Figure 194.

4 Click **Disabled - Pass all**, **Enabled - Do not include IP phones**, or **Enabled - Include IP phones**.

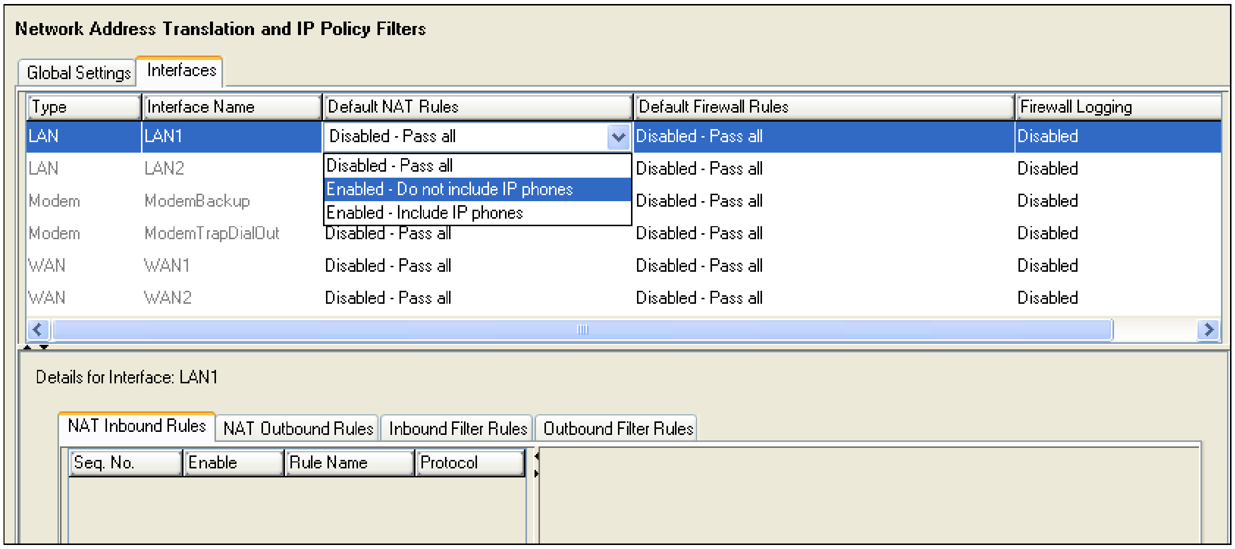
If you choose **Enabled - Include IP phones**, the NAT default rules apply to all data traffic including IP telephony traffic.
There are three default rules set. The first rule is for outbound TCP/UDP traffic. The second rule is for outbound IP traffic. The third rule is for inbound TCP/UDP traffic on ports 7000, 7001, and 7002. The IP address for the Public address is the IP address of the interface you configure. The system automatically fills in the rule order. If you choose to add additional rules, the default rules still remain.

If you choose **Enabled - do not include IP phones**, the NAT default rules do not apply to IP telephony traffic, but do apply to all other traffic.
If you choose this option, there are two default rules set. One is for outbound TCP/UDP traffic. The other is for outbound IP traffic. The IP address for the Public address is the IP address of the interface you configure. The system automatically fills in the rule order. If you choose to add additional rules, the default rules still remain.

If you choose **Disabled**, the Default Rules are removed.
The default is Disabled.

Note: The default rules are only for traffic initiated in the outbound direction. You must add rules for inbound traffic or packets will pass in without translation.

Figure 194 Default NAT Rules



Type	Interface Name	Default NAT Rules	Default Firewall Rules	Firewall Logging
LAN	LAN1	Disabled - Pass all	Disabled - Pass all	Disabled
LAN	LAN2		Disabled - Pass all	Disabled
Modem	ModemBackup		Disabled - Pass all	Disabled
Modem	ModemTrapDialOut	Disabled - Pass all	Disabled - Pass all	Disabled
WAN	WAN1	Disabled - Pass all	Disabled - Pass all	Disabled
WAN	WAN2	Disabled - Pass all	Disabled - Pass all	Disabled

Disabled - Pass all
Enabled - Do not include IP phones
Enabled - Include IP phones

Details for Interface: LAN1

Seq. No.	Enable	Rule Name	Protocol

Adding a Rule to an interface

Note: The maximum number of Rules you can add is 32.

1 Click **Configuration > Data Services > NAT and Filters > Interfaces** tab.

2 Click the interface you want to configure.
The details panel appears.

3 Click the **NAT Inbound Rules Tab.**

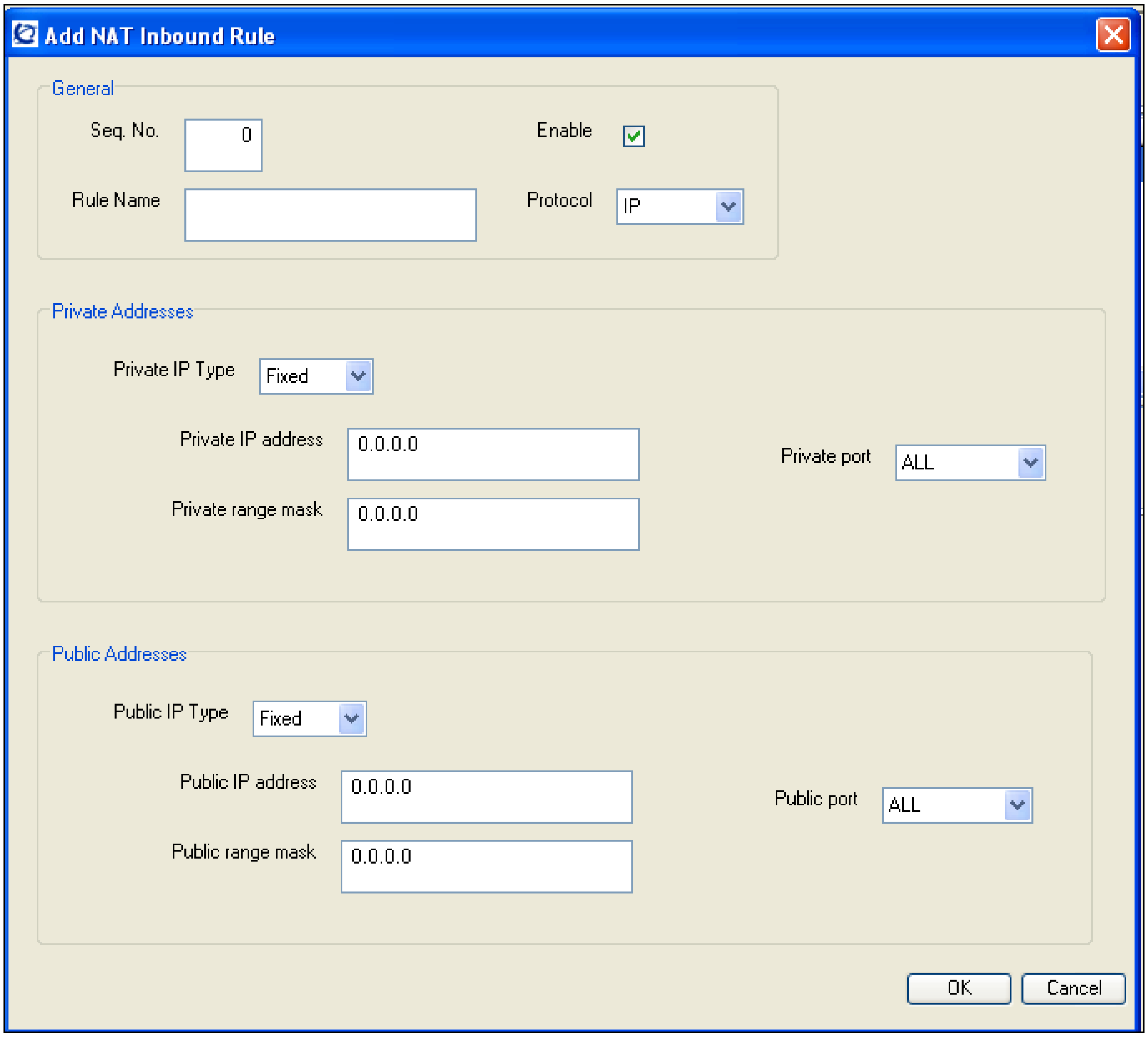
Note: Follow this procedure to add an outbound rule.

4 Click **Add**.
The **Add NAT Inbound Rule** panel appears. See Figure 195.

5 Configure the Rule settings. Refer to the information in Table 23.

6 Click **OK**.

Figure 195 Adding a NAT Inbound Rule



Add NAT Inbound Rule

General

Seq. No. 0

Enable ☑

Rule Name

Protocol IP

Private Addresses

Private IP Type Fixed

Private IP address 0.0.0.0

Private port ALL

Private range mask 0.0.0.0

Public Addresses

Public IP Type Fixed

Public IP address 0.0.0.0

Public port ALL

Public range mask 0.0.0.0

Table 23 NAT Inbound Rule Settings (Sheet 1 of 3)

Attribute	Value	Description
General		
Seq. No.	<numeric>	Determines the rule order. Choose the position of the rule by providing the sequence number value. Previous rules will be shifted accordingly. To change the order of a rule, use the modify button, and select the new sequence number value to use in the modify window. Previous rules will be shifted accordingly.
Rule Name	<text box>	
Enable	<check box>	Determines if a rule is active in the list of rules. Default: Selected

Table 23 NAT Inbound Rule Settings (Sheet 2 of 3)

Attribute	Value	Description
Protocol	IP TCP UDP TCP/UDP	Choose the protocol for this interface.
Private Addresses		
Private IP Type	Fixed Dynamic	Specify the IP type. Use Dynamic when the IP is assigned by an outside source. If you specify Dynamic, Private IP and Private Mask do not need to be entered. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0. Default: Fixed
Private IP Address	<IP address>	Specify the Private IP address. If the Private IP type is fixed, the Rule is invalid without this IP address.
Private range mask	<IP address>	Specify the mask to use with the Private IP. If you want the Rule to apply to a single Private IP address (the Private IP entered), enter 255.255.255.255. If you want the Rule to apply to all Private IP addresses, enter 0.0.0.0.
Private port	<drop-down list>	Specify a single entry for one of the following: ALL, FTP, Telnet, SMTP, SNP, SNMP-TRAP, DNS, TFTP, Gopher, Finger, H.323, SIP, HTTP, HTTPS, POP3, NNTP, SUNRPC, SUNNFS, UNISTIM, or CUSTOM. **Note**: If CUSTOM is used, Private Start Port and Private End Port must be set.
Public Addresses		
Public IP Type	Fixed Dynamic	Specify the IP type. Use Dynamic when the IP is assigned by an outside source. For example, your Internet Service Provider (ISP) assigns your IP address. If you specify Dynamic, Public IP and Public Mask do not need to be entered. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0. Default: Fixed
Public IP address	<IP address>	Specify the Public IP address. This address should be on the outside network. **Note**: If you may want to use an address from the internal network, refer to "Configuring NAT to change the source IP address used for WAN traffic" on page 614.
Public range mask	<IP address>	Specify the mask to use with the Public IP. If you want the Rule to apply to a single Public IP address (the Public IP entered), enter 255.255.255.255. If you want the Rule to apply to all Public IP addresses, enter 0.0.0.0.

Table 23 NAT Inbound Rule Settings (Sheet 3 of 3)

Attribute	Value	Description
Public port		Specify a single entry for one of the following; ALL, FTP, Telnet, SMTP, SNMP, SNMP-TRAP, DNS, TFTP, Gopher, Finger, H.323, SIP, HTTP, HTTPS, POP3, NNTP, SUNRPC, SUNNFS, UNISTIM, or CUSTOM. **Note**: If CUSTOM is used, Public start port and Public end port must be set
Action		
OK	<button>	Click to add entry.
Cancel	<button>	Click to cancel entry.

Note: If you do not configure the public and private masks correctly, mappings to non-existent systems can occur. You must specify both Private and Public addresses that exist on the BCM. For example, if you configure an outbound rule, the Public IP address and Public Mask are the translated addresses. These addresses must be assigned or packets will be sent to a non-existent destination. For inbound rules, the translated address is the Private Address and Mask.

If you want the rule to apply to one IP address only, you must enter a Mask of 255.255.255.255. If you enter any other Mask, the rule will apply to more than one IP address.

Modifying a Rule to an Interface

1 Click **Configuration > Data Services > NAT and Filters > Interfaces** tab.

2 Click the interface you want to configure.
The details panel appears.

3 Click the **NAT Inbound Rules Tab** or **NAT Outbound Rules Tab.**

4 Select the rule to modify.

5 Click the **Modify** button.
The **Modify NAT Inbound/Outbound Rule panel** appears.

6 Modify the Rule settings.

7 Click **OK**.

Deleting a Rule to an Interface

1. Click **Configuration > Data Services > NAT and Filters > Interfaces** tab.
2. Click the interface you want to configure.
   The details panel appears.
3. Click the **NAT Inbound Rules Tab** or **NAT Outbound Rules Tab.**
4. Select the rule to delete.
5. Click **Delete**.
   A message appears that asks you to confirm the deletion.
6. Click **Yes**.

Configuring NAT to change the source IP address used for WAN traffic

Normally, when traffic originates from the BCM, the source IP address of the data packets is the IP address of interface that is on the edge. For example, if the traffic is sent out of the WAN1 interface, the source IP address of the data packets is the IP address of WAN1.

You can use NAT rules to change the source IP address of the data packets to one of the other data network interfaces. To create this type of NAT rule, you must comply with the following rules:

- The IP addresses you enter must be addresses that can be correctly routed.
- The IP address you enter in the **Public IP** box must be the IP address of one of the other data interfaces (for example, LAN1 or LAN2).
- These NAT rules are only available on the WAN interface and the ISDN WAN backup interface. They are not available on the V.92 WAN backup interface.

Chapter 67

Configuring IP Filter Rules

The BCM IP Filter Rules feature is one of the security features BCM offers to protect networks against intruders and to optionally provide DiffServ Marking. The security and IP Filters features include IP Firewall functionality and are also used for controlling what outside resources users are able to access.

The following path indicates where to access the Filter Rule settings in Element Manager:

Element Manager: **Configuration > Data Services > NAT and Filters**

Refer to the following topics:

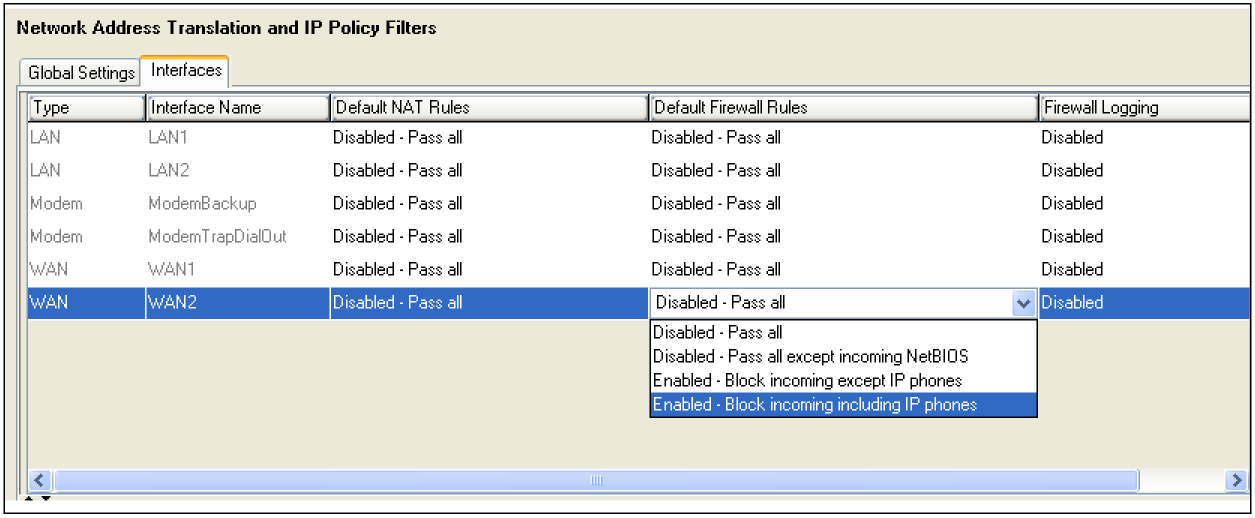
- Default Rules
- "Inbound Rules" on page 619
- "Outbound Rules" on page 626

Default Rules

Default rules are provided which block all connections for inbound requests and allow all connections for outbound requests.

Note: When IP policy filters are globally enabled, any default rule that allows traffic is considered stateful. If you globally disable IP policy filters, all stateful rules are flushed immediately.

Figure 196 Default Firewall Rules



Network Address Translation and IP Policy Filters

Type	Interface Name	Default NAT Rules	Default Firewall Rules	Firewall Logging
LAN	LAN1	Disabled - Pass all	Disabled - Pass all	Disabled
LAN	LAN2	Disabled - Pass all	Disabled - Pass all	Disabled
Modem	ModemBackup	Disabled - Pass all	Disabled - Pass all	Disabled
Modem	ModemTrapDialOut	Disabled - Pass all	Disabled - Pass all	Disabled
WAN	WAN1	Disabled - Pass all	Disabled - Pass all	Disabled
WAN	WAN2	Disabled - Pass all	Disabled - Pass all	Disabled

Disabled - Pass all
Disabled - Pass all except incoming NetBIOS
Enabled - Block incoming except IP phones
Enabled - Block incoming including IP phones

Caution: If you enable the Default Rules on the interface used by Element Manager to access the BCM, then you will lose connectivity with the BCM unless the user configures the rules described in the section "Accessing Element Manager through the Firewall" on page 638.

To enable default rules

1 Click **Configuration > Data Services > NAT and Filters > Interfaces tab**.

2 Double-click a field under the **Default Firewall Rules** column.
A drop-down list appears.

3 Click one of the following options:

- **Disabled - Pass all**
  The IP Firewall does not check the traffic on this interface. Therefore, all traffic on this interface, both incoming and outgoing, is passed through.
- **Disabled - Pass all except incoming NetBIOS**
  The IP firewall allows outbound and inbound traffic, except that it blocks inbound NetBIOS messages. BCM creates three read-only inbound IP filter rules on behalf of the user.
- **Enabled - Block incoming except IP phones**
  Allows Nortel's IP telephony signaling traffic (for example UNISTIM) through, but blocks all other traffic on this interface.

Note: You must still specify an inbound IP Filter rule to allow either H.323 or SIP signaling traffic. This setting only allows the IP telephone to contact the system to register.

Also, Registration must be turned on under **Configuration > Resources > Telephony Resources > IP Terminal Global Settings tab**, before the telephone can access the system to register.

- **Enabled - Block incoming including IP phones**
  Blocks all traffic on this interface that is initiated from the inbound direction, including IP telephony traffic.

The default is **Disabled - Pass all**.

Note: Setting an Inbound Rule that blocks all incoming packets and disabling the Default Rules is not the same as enabling the Default Rules.

When block all incoming packets and disable the Default Rules is selected, packets that originate from inside the Firewall are not treated as Stateful. When a response packet is returned, it will not match the Inbound Rule and will be blocked.

When you enable the Default Rules, packets that originate from inside the Firewall are treated as Stateful. When a response packet is returned, it will match the existing state and will be passed.

Configuring IP Filters for an interface

The following describes how to enable and configure IP Filters.

Refer to the following information:

- "To add an inbound filter" on page 620
- "To modify an inbound filter" on page 621
- "To delete an inbound filter" on page 621
- "To add an outbound filter" on page 636
- "To modify an outbound filter" on page 636
- "To delete an outbound filter" on page 636
- "Configuring the order of the inbound and outbound filters for an interface" on page 637

To enable IP filters

1. Click **Configuration > Data Services > NAT and Filters > Global Settings** tab. See Figure 197.
2. Select the **Enable IP Policy Filters** check box.

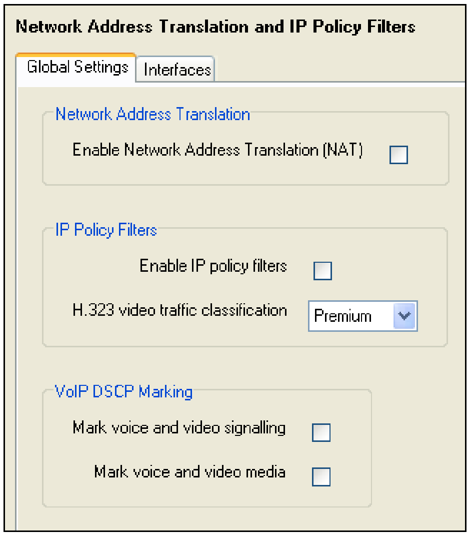
For outbound rules configure the following:

Refer to Table 24 for an explanation of these fields.

1. Select the quality level of the video classification from the **H.323 Video traffic classification** drop-down list.
2. Enable or disable the **Mark voice and video signalling** check box.
3. Enable or disable the **Mark voice and video media** check box.

Note: You can configure rules several ways; using default rules, setting up individual rules, or a combination of the two.

Figure 197 IP Policy Filters and VoIP DSCP Marking



Network Address Translation and IP Policy Filters

Network Address Translation

Enable Network Address Translation (NAT) □

IP Policy Filters

Enable IP policy filters □

H.323 video traffic classification: Premium

VoIP DSCP Marking

Mark voice and video signalling □

Mark voice and video media □

Table 24 Global Settings (Sheet 1 of 2)

Attribute	Value	Description
IP Policy Filters		
Enable IP policy filters	<check box>	Selected - IP filters are enabled Cleared - IP filters are disabled. Default: Cleared.
H.323 video traffic classification	Premium Best Effort	Select the DiffServ Code Point Per-Hop Behavior that is used for H.323 Video traffic. When Premium is selected, the Expedited Forwarding DSCP PHB (hexadecimal value B8) is marked on the IP datagram. When Best Effort is selected, the DeFault (DF) DSCP PHB (value 0) is marked on the IP datagram. **Note:** Choose Best Effort if you want to prevent IP Video traffic from competing with IP Telephony traffic. Default: Premium
VoIP DSCP Marking		

Table 24 Global Settings (Sheet 2 of 2)

Attribute	Value	Description
Mark voice and video signalling	<check box>	Remark voice and video signaling packets for pass-through and locally generated traffic. If selected, DSCP value CS5 is marked for the following packets: SIP signaling Unistim signaling H.225 gatekeeper RAS signaling H.225 signaling T120 signaling H.245 signaling Default: Cleared
Mark voice and video media	<check box>	Remark voice and video media packets for pass-through and locally generated traffic. If selected, the DSCP values below are marked for the following packets: H.323 and SIP media uses DSCP value EF. H.323 video uses DSCP value DF only if the H.323 Video Traffic Classification field is set to Best Effort. Default: Cleared

Inbound Rules

The BCM IP Filters feature is one of the security features BCM offers to protect your network against intruders. The security and firewall features are also used for controlling what outside resources your users will be able to access.

The following features are part of the BCM firewall:

- "Basic (stateless) Packet Filter" on page 620
- "Stateful Packet Filters" on page 715

Also refer to:

- "Accessing Element Manager through the Firewall" on page 638
- "Firewall rules for BCM with Dialup interfaces" on page 639

Caution: When blocking incoming packets, make sure you do not block your access to Element Manager on the system. See "Accessing Element Manager through the Firewall" on page 638 for a description of the rules.

Packet filtering

A packet filter is a firewall facility that inspects incoming and outgoing packets and uses this information to determine which network packets to allow through the firewall. The traffic may or may not be tracked by keeping the state of the connection.

Basic (stateless) Packet Filter

BCM supports basic (or stateless) packet filtering for IP protocols. Stateless packet filtering examines each packet and determines whether or not to pass it through based on the rules entered. No state is maintained for packets evaluated using stateless rules.

Basic Packet Filters are configured by clearing the **Stateful** check box on the **Add** or **Modify** filter rules dialog boxes.

For more information, refer to "Stateful Packet Filters" on page 715.

IP Filters and NAT

When you use NAT and IP Filters, there are two interactions you need to be aware of.

- On inbound traffic, the NAT rules are applied before the IP Firewall Filter rules.
- On outbound traffic, the IP Firewall Filter rules are applied before the NAT rules.

Adding an Inbound filter rule

Note: Add the rule in Table 160 to ensure access to the Element Manager is not blocked.

To allow access for SSH, you must set the filters to allow the flow of packets to the SSH port (port 22).

To allow access for Telnet, you must set the filters to allow the flow of packets to the Telnet port (port 23).

To allow access for FTP, you must set the filters to allow the flow of packets to the FTP port (port 20)

Note: To mark inbound packets the stateful check box must be selected. Refer to "Stateful Packet Filters" on page 715 for more information.

To add an inbound filter

1. Click the **Inbound Filter Rule** tab.
   The Inbound Filter Rule details panel appears.
2. Click **Add.**
   The **Add Inbound Filter Rule** dialog box appears. See Figure 198.
3. Configure the Inbound Filter Rule settings. Refer to the information in Table 156.
4. Click **OK**.

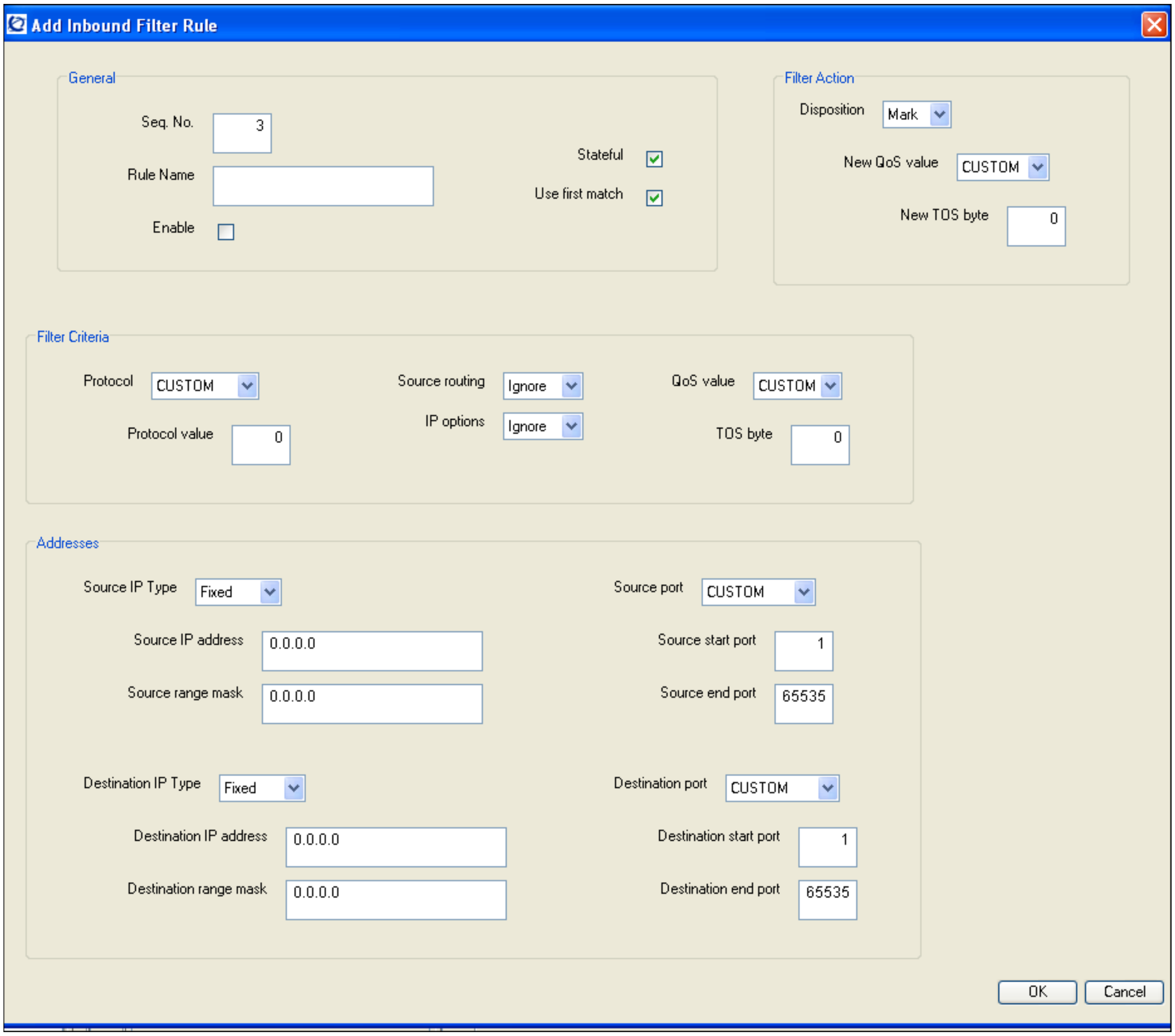
To modify an inbound filter

1. Click the **Inbound Filter Rule** tab.
   The Inbound Filter Rule details panel appears.
2. Select the Inbound Filter you want to modify.
3. Click **Modify**.
4. Modify the Inbound Filter Rule attributes.
5. Click **OK**.

To delete an inbound filter

1. Click the **Inbound Filter Rule** tab.
   The Inbound Filter Rule details panel appears.
2. Select the Inbound Filter you want to delete.
3. Click **Delete**.
   A message appears that asks you to confirm the deletion.
4. Click **Yes**.

Figure 198 Add Inbound Filter Rule



Add Inbound Filter Rule

General

Seq. No. 3

Rule Name

Enable □

Stateful ☑

Use first match ☑

Filter Action

Disposition Mark

New QoS value CUSTOM

New TOS byte 0

Filter Criteria

Protocol CUSTOM

Protocol value 0

Source routing Ignore

IP options Ignore

QoS value CUSTOM

TOS byte 0

Addresses

Source IP Type Fixed

Source IP address 0.0.0.0

Source range mask 0.0.0.0

Source port CUSTOM

Source start port 1

Source end port 65535

Destination IP Type Fixed

Destination IP address 0.0.0.0

Destination range mask 0.0.0.0

Destination port CUSTOM

Destination start port 1

Destination end port 65535

Table 156 Add Inbound Filter Rule (Sheet 1 of 5)

Attribute	Value	Description
General		
Seq. No.	<numeric>	Set the rule order. In the “Add” window, you can choose the position of the rule by providing the sequence number value. The new rule will be inserted at the position determined by the sequence number, and the previous rules will be shifted accordingly. In the “Modify” window, the user can change the sequence number. The rule will be moved at the position determined by the sequence number, and the previous rules will be shifted accordingly.

Table 156 Add Inbound Filter Rule (Sheet 2 of 5)

Attribute	**Value**	**Description**
Rule Name		Assign a name to the rule. The maximum length is 15 characters. This field is optional and can be left empty. The same rule name can be repeated under the same interface.
Enable	<check box>	Determine if a rule is active in the list of rules. Default**: selected**.
Stateful	<check box>	Specify if the states of connections that match this rule will be monitored. This permits the creation of one-way rules. For example, you can permit inside traffic to return but block traffic originating from the outside. **Note:** Be aware of the limitation of stateful sessions with VoIP DSCP marking rules. For VoIP DSCP marking to work, the user can either configure an outbound filter rule (stateful or not) with disposition to “mark” or create an inbound stateful filter rule with disposition to “mark”. A disposition of “mark” allows the inbound packet to pass but does not mark it. Default: **selected**.
Use first match	<check box>	Specify if the first rule match is used. If cleared, the last rule match is used. If multiple rules match, either the first rule with this check box selected is used, or the last rule that matches in the list of available matching rules is used if all check boxes are cleared, there is no concept of more specific matches or less specific matches other than being determined by the order of the rule in the list of rules. The lowest sequence number is looked up first. Default: **Selected**.
Filter Action		
Disposition	Block Pass Mark	Specify if a packet that matches this rule is blocked or passes through. Default: **Block**.
New QoS value	EF, CS7, CS6, CS5, CS4, CS3, CS2, CS1, AF41, AF42, AF43, AF31, AF32, AF33, AF21, AF22, AF23, AF11, AF12, AF13, DF(CS0), or CUSTOM.	If the Disposition attribute is set to Mark, select the new DiffServ Code Point (DSCP) value that will put on the IP datagram. If the Disposition attribute is set to either Block or Pass, this attribute value is ignored. **Note:** The DSCP value is the 6-bit field of the ToS byte. Default: **DF**. This value is referred to as DeFault DSCP PHB (value 0).
New ToS byte	<0-255>	Specify a custom ToS byte value not available in the DSCP values. This value applies to the 8-bit field of the ToS byte. This attribute can only be set if New QoS Value is set to **CUSTOM**. **Note:** This value applies to the 8-bit field of the ToS byte. Default: 0.

Table 156 Add Inbound Filter Rule (Sheet 3 of 5)

Attribute	Value	Description
Filter Criteria		
Protocol	TCP, UDP, TCP/UDP, ICMP, OSPF, IPSEC_AH, IPSEC_ESP, IGNORE, or CUSTOM	Specify the protocol type of the packet to be filtered. If you choose **CUSTOM**, the value provided in the Protocol Value attribute will be used. Default: **IGNORE**.
Protocol value	<0 to 255>	Specify a custom protocol value not available in the Protocol attribute. This attribute can only be set if Protocol is set to **CUSTOM**.
Source routing	Present Absent Ignore	Specify how the Source Routing is checked. **Present**: Rule matches only if the packet has the source routing option set. **Absent**: Rule matches only if the packet does not have the source routing option set. **Ignore**: The source routing option in the packet is not checked and therefore all packets will match. Default: **Ignore**.
IP options	Present Absent Ignore	Specify how the IP Options are checked. **Present**: Rule matches only if the packet has the IP options set. **Absent**: Rule matches only if the packet does not have the IP options set. **Ignore**: The IP options in the packet are not checked and therefore all packets will match. Default: **Ignore.**
QoS value	EF, CS7, CS6, CS5, CS4, CS3, CS2, CS1, AF41, AF42, AF43, AF31, AF32, AF33, AF21, AF22, AF23, AF11, AF12, AF13, DF(CS0), None, or CUSTOM	Select a DiffServ Code Point (DSCP) value to match the filter from the list. If you choose **CUSTOM**, the value provided in the ToS Byte attribute will be used. **Note:** The DSCP value is the 6-bit field of the ToS byte. Default: **None**.
ToS byte	<0-255>	Specify a custom ToS byte value not available in the DSCP attribute. **Note:** This value applies to the 8-bit field of the ToS byte. This attribute can only be set if QoS Value is set to **CUSTOM**. Default: 0
Addresses		
Source IP Type	Fixed Dynamic	Specify if the Source IP is **Fixed** or **Dynamic**. Use **Dynamic** when the IP is assigned by an outside source. For example, your Internet Service Provider (ISP) assigns your IP address. If you specify Dynamic, Source IP and Source IP Mask do not need to be entered. The default is **Fixed**. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0.

Table 156 Add Inbound Filter Rule (Sheet 4 of 5)

Attribute	Value	Description
Source IP address	<IP address>	Specify the source address of the packet to be filtered. The default is 0.0.0.0.
Source range mask	<IP address>	Specify the source address mask of the packet to be filtered. If you enter 255.255.255.255, then the Source IP is a single address. If you enter 0.0.0.0, then the Source IP is all possible addresses.
Source port	ALL, OAM, FTP, Telnet, SMTP, SNMP, SNMP-TRAP, DNS, DHCP, TFTP, Gopher, Finger, HTTP, HTTPS, H.323, SIP, POP, NNTP, NetBIOS, SUNRPC, SUNNFS, DCOM, and CUSTOM	Specify a single entry for the source port. You can specify a custom value or a range by choosing **CUSTOM**. If **CUSTOM** is used, Source Start Port and Source End Port must be set.
Source start port	<1-65535>	Specify the start of a port range when Source Port is set to **CUSTOM**. The value must be less or equal to Source End Port.
Source end port	<1-65535>	Specify the end of a port range when Source Port is set to **CUSTOM**. The value must be greater or equal to Source Start Port.
Destination IP Type	Fixed Dynamic	Specify if the Destination IP Type is **Fixed** or **Dynamic**. Use Dynamic when the IP is assigned by an outside source. For example, your Internet Service Provider (ISP) assigns your IP address. If you specify Dynamic, Destination IP and Destination IP Mask do not need to be entered. Default: **Fixed**. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0.
Destination IP address	<IP address>	Specify the Destination IP address. Default: 0.0.0.0.
Destination range mask	<IP address>	Specify the destination address mask of the packet to be filtered. If you enter 255.255.255.255, then the Destination IP is a single address. If you enter 0.0.0.0 then the Destination IP is all possible addresses.
Destination port	ALL, OAM, FTP, Telnet, SMTP, SNMP, SNMP-TRAP, DNS, DHCP, TFTP, Gopher, Finger, HTTP, HTTPS, H.323, SIP, POP, NNTP, NetBIOS, SUNRPC, SUNNFS, DCOM, and CUSTOM	Select the destination port. You can specify a custom value or a range by choosing **CUSTOM**. If **CUSTOM** is used, Destination Start Port and Destination End Port must be set.

Table 156 Add Inbound Filter Rule (Sheet 5 of 5)

Attribute	Value	Description
Destination start port	<1-65535>	Specify the start of a port range when Destination Port is set to **CUSTOM**. The value must be less or equal to Destination End Port.
Destination end port	<1-65535>	Specify the end of a port range when Destination Port is set to **CUSTOM**. The value must be greater or equal to Destination Start Port.

Outbound Rules

Rules are created to block, pass, or pass and mark based on the following information: source IP address, source mask, source port, destination IP address, destination mask, destination port, protocol, ToS byte, packet direction, source routing, and IP header options.

A packet is processed as follows:

1. The BCM receives, or generates, a packet.
2. Filters are applied.
3. Filter actions are taken and the packet can be modified (DSCP).
4. The packet is assigned a QoS class. A QoS class is designated using the DSCP values and the packet is placed in the appropriate egress queue according to its priority marking as described above.
5. The queues are serviced in a round-robin fashion (strict priority or HTB).

Traffic monitoring overview

BCM enables system administrators to implement classes of service and assign priority levels to different types of traffic. Using Element Manager, you can configure rules that monitor the characteristics of traffic (for example, its source, destination, and protocol) and perform a controlling action on the traffic when certain user-defined characteristics are matched.

Refer to:

- Differentiated Services (DiffServ) overview
- “DiffServ components” on page 628
- “IP service classes” on page 628

Differentiated Services (DiffServ) overview

Differentiated services (DiffServ) is a Quality of Service (QoS) network architecture that offers varied levels of service for different types of data traffic. DiffServ allows you to designate a specific level of performance on a packet-by-packet basis instead of using the “best-effort” model for your data delivery. You can give preferential treatment (prioritization) to applications that require high performance and reliable service, such as voice and video over IP.

BCM includes the capability to enhance your network traffic management. For each packet, there is an octet in the packet header, the DiffServ (DS) field, that you can designate for specific service. For IP packets, six bits of the DiffServ field is the DiffServ Code Point (*DSCP)*. The DSCP value defines how the packet is to be treated as it travels through the network. You can set traffic criteria to match the DS field, and rule actions to change the DiffServ field to conform to various other mappings.

DiffServ IP Quality of Service (QoS) architecture

DiffServ uses a simple mechanism that relies on a special encoding of the first 6 bits of the DiffServ byte in the IP header. This byte is the IPv4 Type of Service (ToS) byte; for IPv6, is the Traffic Class byte. The first 6 bits of this byte are called the DiffServ Code Point (DSCP).

In the packet forwarding path, differentiated services are processed by mapping the packet DSCP to a particular forwarding treatment, or per hop behavior (PHB), at each network node along its path. The code points may be chosen from a set of 32 standard values, a set of 16 recommended values to be used in the future, or a set of 16 values reserved for experimentation and local use. Of the 32 standard values, there are 8 Class Selector code points that are used primarily (but not exclusively) for backward compatibility with existing definitions of the ToS byte.

BCM is a DiffServ node that can support DiffServ functions and behavior. DiffServ architecture defines a DiffServ-capable domain as a contiguous set of DiffServ-compliant nodes that operate with a common set of service provisioning policies and PHB definitions. The DiffServ domain is an autonomous system or network such as an internet service provider (ISP) network or campus LAN.

DiffServ assumes the existence of a service level agreement (SLA) between DiffServ domains that share a border. The SLA defines the profile for the aggregate traffic flowing from one network to the other based on rule criteria. In a given traffic direction, the traffic is expected to be shaped at the egress point of the upstream network and policed at the ingress point of the downstream network.

End-to-end QoS is enabled, typically through bilateral agreements (an agreement between two DiffServ domains), between all the domains from the sender to the receiver. These agreements aid in consistent PHB and QoS performance across all domains.

Typically, there are three types of edge devices in a DiffServ domain:

- Edge node (EN) — the switch or router connected directly to the desktop end station (ES) (BCM is an edge node in the DiffServ domain)
- Ingress border node (IBN) — the ingress router at the boundary between two DiffServ domains
- Egress border node (EBN) — the egress router at the boundary between two DiffServ domains

DiffServ components

The DiffServ architecture is comprised of the following components:

- Traffic conditioners — These components include classifiers, DiffServ-byte markers, shapers, policiers and profilers. Marking is performed at network boundaries, including the edges of the network (first hop router or switch or source host) and administrative boundaries between networks or autonomous systems. Traffic conditions should exist at DiffServ ingress and egress nodes. BCM is an edge switch that supports packet classification based on header information in layer 3 and layer 4 of the Open System Interconnection (OSI) layering model. BCM can mark and re-mark IP traffic based on the rules you define.
- Packet schedulers and queue managers — PHBs are expected to be implemented by employing a range of queue service and/or queue management disciplines on a network node output interface queue (for example, HTB or drop preference queue management). DiffServ does not require a particular discipline for queue management or servicing to realize a particular service. All DiffServ nodes should support the packet scheduling and queue management algorithms that are necessary to implement the required PHB.

  BCM supports a queue service discipline that allows packets to be serviced in an absolute priority fashion or using a weighted fair queueing scheduler. This service discipline ensures that packets in the highest-priority queue are serviced quickly without starving lower-priority queues.
- Bandwidth brokers (not supported in BCM) — Bandwidth brokering is responsible for bandwidth allocation, QoS rule management, and flow admission control in a given DiffServ domain. BCM does not support bandwidth brokering or traffic admission control.

IP service classes

BCM supports the following service classes:

- Premium class is an end-to-end service functioning similarly to a virtual leased line. Traffic in this service class is guaranteed an agreed upon peak bandwidth. Traffic requiring this service should be shaped at the network boundary in order to undergo a negligible delay and delay variance. This service class is suitable for real time applications like video and voice over IP. The recommended PHB for this service is the Expedited Forwarding (EF) PHB.
- Standard Network Control Class is used for network control traffic and has priority over user traffic.
- Platinum, Gold, Silver, and Bronze classes use the Assured Forwarding PHB. These classes are used for real time, delay tolerant traffic and non-real time, mission critical traffic.
- Best Effort (standard) class is the standard Internet packet service with an additional, optional use of traffic profiling that is used at the network boundary to request a better effort treatment for packets that are in-profile (packets that do not break the service agreements between the user and the service provider).

Table 157 describes the service classes and the required treatment. The table shows how the service classes are mapped to the BCM queues.

Table 157 Service classes

Traffic category	Service class	Application type	Required treatment
Real time, delay intolerant, fixed bandwidth	Premium	Person to person communications requiring interaction (such as VoIP).	Absolute bounded priority over user traffic. No packet loss for in-profile traffic. Virtual leased line with lowest amount of latency. Provisioned for peak rate.
Standard Network Control	Network	Standard network control traffic	Priority over user traffic. Guaranteed minimum bandwidth
Real time, delay tolerant, low variable bandwidth	Platinum	Person to person communications requiring interaction with additional minimal delay (such as low cost VoIP).	Higher-priority scheduling providing guaranteed minimum provisioned bandwidth. Competes for additional bandwidth.
Real time, delay tolerant, high variable bandwidth	Gold	Single human communication with no interaction (such as Web site streaming video).	High-priority scheduling providing guaranteed minimum provisioned bandwidth. Competes for additional bandwidth.
Non-real time, mission critical, interactive	Silver	Transaction processing (such as Telnet, Web browsing).	Medium priority scheduling providing guaranteed minimum provisioned bandwidth. Competes for additional bandwidth.
Non-real time, mission critical, non-interactive	Bronze	For example, E-mail, FTP, SNMP.	Lower-priority scheduling providing guaranteed minimum provisioned bandwidth. Competes for additional bandwidth.
Non-real time, non-mission critical	Standard	Bulk transfer (such as large FTP transfers, after-hours tape backup).	Best effort delivery. Uses remaining available bandwidth.

These Required treatments (or service class behaviors) for these Service classes are implemented using seven queues and a scheduler for these queues. Queue 1 has the highest priority, referred to as Strict Priority. Queues 2 to 7 are scheduled according to a HTB (Hierarchical Token Buckets) scheme. This scheduler allows each service class to use a minimum traffic rate, and allows unused bandwidth of other classes in the HTB scheduler to be redistributed to other classes. With this scheduler, you can arrange for the premium service class to behave as a strict priority scheduler and provide shaping capabilities to prevent starving the other six queues. Table 158 summarizes the mappings between service classes, queues and DSCP codes.

Table 158 Default Queue mapping for BCM

NSC	Default DSCP	BCM Queue	BCM Scheduler
Premium	EF, CS5	1	Strict Priority
Network	CS7, CS6	2	HTB
Platinum	AF4x, CS4	3	HTB
Gold	AF3x, CS3	4	HTB
Silver	AF2x, CS2	5	HTB
Bronze	AF1x, CS1	6	HTB
Standard	DF (CS0) All unspecified DSCPs	7	HTB

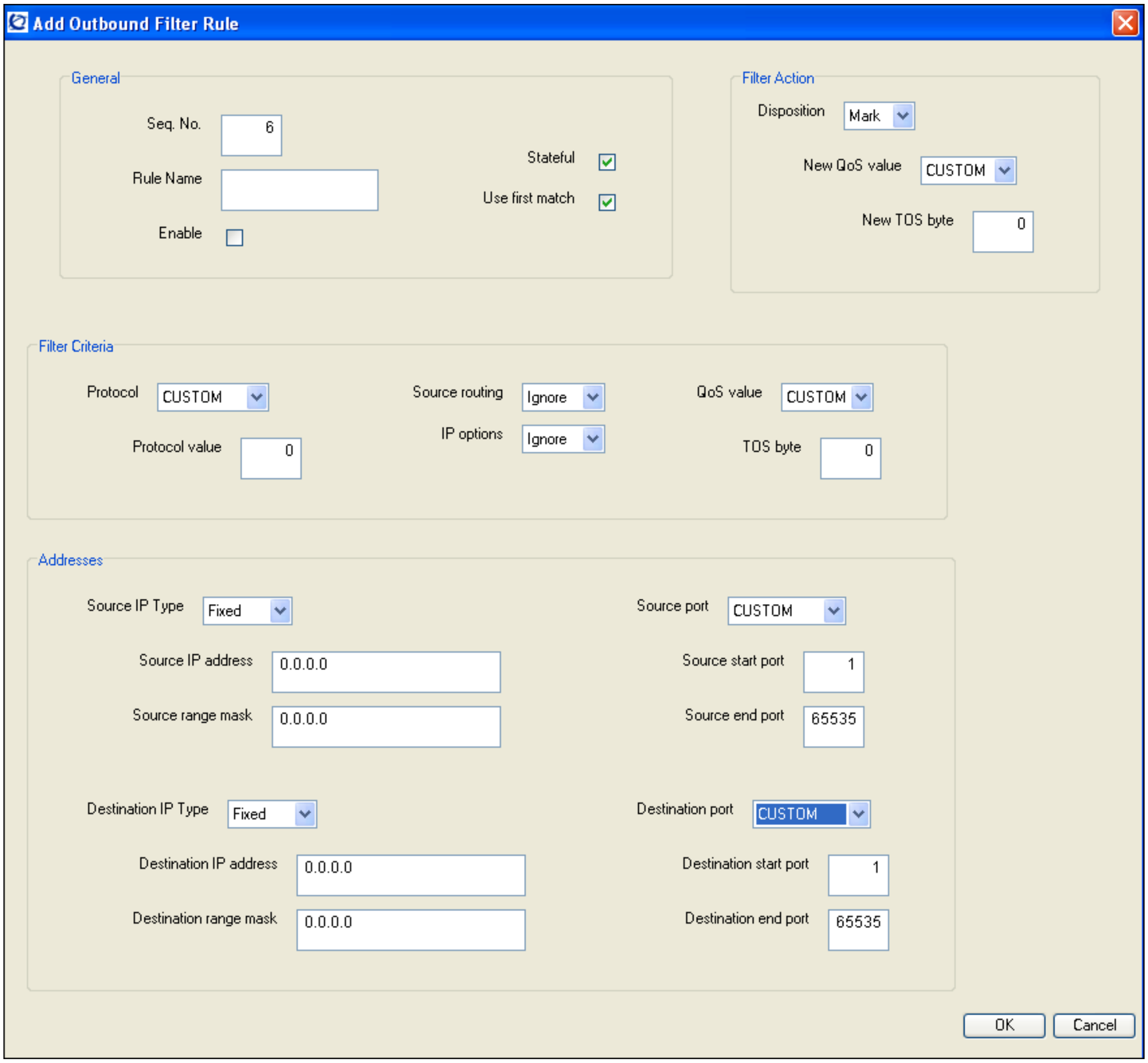
Implementing Quality of Service (QoS)

The QoS application delivers a set of tools that, when optimally configured, combat escalating bandwidth costs and optimize application performance in your network.

QoS tools allow you to prioritize your critical applications and sensitive traffic. You can tailor appropriate services to support this traffic over the wide area, thus maintaining the necessary performance levels on an end-to-end basis. The *BCM 4.0 Administration Guide* (N0060598) provides information on QoS metrics.

To configure QoS, refer to "Configuring QoS Queuing" on page 687.

Figure 199 Add Outbound Filter Rule



Add Outbound Filter Rule

General

Seq. No. 6

Rule Name

Enable □

Stateful ☑

Use first match ☑

Filter Action

Disposition Mark

New QoS value CUSTOM

New TOS byte 0

Filter Criteria

Protocol CUSTOM

Protocol value 0

Source routing Ignore

IP options Ignore

QoS value CUSTOM

TOS byte 0

Addresses

Source IP Type Fixed

Source IP address 0.0.0.0

Source range mask 0.0.0.0

Source port CUSTOM

Source start port 1

Source end port 65535

Destination IP Type Fixed

Destination IP address 0.0.0.0

Destination range mask 0.0.0.0

Destination port CUSTOM

Destination start port 1

Destination end port 65535

Table 159 Add Outbound Filter Rule (Sheet 1 of 4)

Attribute	Value	Description
General		
Seq. No.	<numeric>	Set the rule order. In the “Add” window, you can choose the position of the rule by providing the sequence number value. The new rule will be inserted at the position determined by the sequence number, and the previous rules will be shifted accordingly. In the “Modify” window, the user can change the sequence number. The rule will be moved at the position determined by the sequence number, and the previous rules will be shifted accordingly.
Rule Name		Specify a name for the rule. The maximum length is 15 characters. This field is optional and can be left empty. The same rule name can be repeated under the same interface.
Enable	<check box>	Determine if a rule is active in the list of rules. Default**: selected**.
Stateful	<check box>	Specify if the states of connections that match this rule will be monitored. This permits the creation of one-way rules. For example, you can permit inside traffic to return but block traffic originating from the outside. **Note:** Be aware of the limitation of stateful sessions with VoIP DSCP marking rules. For VoIP DSCP marking to work, the user can either configure an outbound filter rule (stateful or not) with disposition to “mark” or create an inbound stateful filter rule with disposition to “mark”. A disposition of “mark” allows the inbound packet to pass but does not mark it. Default: **selected**.
Use first match	<check box>	Specify if the first rule match is used. If cleared, the last rule match is used. If multiple rules match, either the first rule with this check box selected is used, or the last rule that matches in the list of available matching rules is used if all check boxes are cleared, there is no concept of more specific matches or less specific matches other than being determined by the order of the rule in the list of rules. The lowest sequence number is looked up first. Default: **Selected**.
Filter Action		
Disposition	Block Pass Mark	Specify if a packet that matches this rule is blocked, passes through, or is DSCP marked and passes through. A value of **Mark** means that the outbound packet is DSCP marked then passes through. This value is allowed for outbound rules (stateful or not) and for stateful inbound rules. A disposition of “mark” allows the inbound packet to pass but does not mark it. Default: **Block**.

Table 159 Add Outbound Filter Rule (Sheet 2 of 4)

Attribute	Value	Description
New QoS value	EF, CS7, CS6, CS5, CS4, CS3, CS2, CS1, AF41, AF42, AF43, AF31, AF32, AF33, AF21, AF22, AF23, AF11, AF12, AF13, DF(CS0), or CUSTOM.	If the Disposition attribute is set to Mark, select the new DiffServ Code Point (DSCP) value that will put on the IP datagram. If the Disposition attribute is set to either Block or Pass, this attribute value is ignored. **Note:** The DSCP value is the 6-bit field of the ToS byte. Default: **DF**. This value is referred to as DeFault DSCP PHB (value 0).
New ToS byte	<0-255>	Specify a custom ToS byte value not available in the DSCP values. This value applies to the 8-bit field of the ToS byte. This attribute can only be set if New QoS Value is set to **CUSTOM**. **Note:** This value applies to the 8-bit field of the ToS byte. Default: 0.
Filter Criteria		
Protocol	TCP, UDP, TCP/UDP, ICMP, OSPF, IPSEC_AH, IPSEC_ESP, IGNORE, or CUSTOM	Specify the protocol type of the packet to be filtered. If you choose **CUSTOM**, the value provided in the Protocol Value attribute will be used. Default: **IGNORE**.
Protocol value	<0 to 255>	Specify a custom protocol value not available in the Protocol attribute. This attribute can only be set if Protocol is set to **CUSTOM**.
Source routing	Present Absent Ignore	Specify how the Source Routing is checked. **Present**: Rule matches only if the packet has the source routing option set. **Absent**: Rule matches only if the packet does not have the source routing option set. **Ignore**: The source routing option in the packet is not checked and therefore all packets will match. Default: Ignore.
IP options	Present Absent Ignore	Specify how the IP Options are checked. **Present**: Rule matches only if the packet has the IP options set. **Absent**: Rule matches only if the packet does not have the IP options set. **Ignore**: The IP options in the packet are not checked and therefore all packets will match. Default: Ignore.
QoS value	EF, CS7, CS6, CS5, CS4, CS3, CS2, CS1, AF41, AF42, AF43, AF31, AF32, AF33, AF21, AF22, AF23, AF11, AF12, AF13, DF(CS0), None, or CUSTOM	Select a DiffServ Code Point (DSCP) value to match the filter from the list. If you choose **CUSTOM**, the value provided in the ToS Byte attribute will be used. **Note:** The DSCP value is the 6-bit field of the ToS byte. Default: None.

Table 159 Add Outbound Filter Rule (Sheet 3 of 4)

Attribute	Value	Description
ToS byte	<0-255>	Specify a custom ToS byte value not available in the DSCP attribute. **Note:** This value applies to the 8-bit field of the ToS byte. This attribute can only be set if QoS Value is set to **CUSTOM**.
Addresses		
Source IP Type	Fixed Dynamic	Specify if the Source IP is **Fixed** or **Dynamic**. Use **Dynamic** when the IP is assigned by an outside source. For example, your Internet Service Provider (ISP) assigns your IP address. If you specify Dynamic, Source IP and Source IP Mask do not need to be entered. The default is **Fixed**. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0.
Source IP address	<IP address>	Specify the source address of the packet to be filtered. The default is 0.0.0.0.
Source range mask	<IP address>	Specify the source address mask of the packet to be filtered. If you enter 255.255.255.255, then the Source IP is a single address. If you enter 0.0.0.0, then the Source IP is all possible addresses.
Source port	ALL, OAM, FTP, Telnet, SMTP, SNMP, SNMP-TRAP, DNS, DHCP, TFTP, Gopher, Finger, HTTP, HTTPS, H.323, SIP, POP, NNTP, NetBIOS, SUNRPC, SUNNFS, DCOM, and CUSTOM	Specify a single entry for the source port. You can specify a custom value or a range by choosing **CUSTOM**. If **CUSTOM** is used, Source Start Port and Source End Port must be set.
Source start port	1-65535	Specify the start of a port range when Source Port is set to **CUSTOM**. The value must be less or equal to Source End Port.
Source end port	1-65535	Specify the end of a port range when Source Port is set to **CUSTOM**. The value must be greater or equal to Source Start Port.
Destination IP Type	Fixed Dynamic	Specify if the Destination IP Type is **Fixed** or **Dynamic**. Use Dynamic when the IP is assigned by an outside source. For example, your Internet Service Provider (ISP) assigns your IP address. If you specify Dynamic, Destination IP and Destination IP Mask do not need to be entered. Default: **Fixed**. **Note**: Dynamic does not match all IP addresses. If you want to match all IP addresses, enter an IP address of 0.0.0.0 and a mask of 0.0.0.0.
Destination IP address	<IP address>	Specify the Destination IP address. Default: 0.0.0.0.

Table 159 Add Outbound Filter Rule (Sheet 4 of 4)

Attribute	**Value**	**Description**
Destination range mask	<IP address>	Specify the destination address mask of the packet to be filtered. If you enter 255.255.255.255, then the Destination IP is a single address. If you enter 0.0.0.0 then the Destination IP is all possible addresses.
Destination port	ALL, OAM, FTP, Telnet, SMTP, SNMP, SNMP-TRAP, DNS, DHCP, TFTP, Gopher, Finger, HTTP, HTTPS, H.323, SIP, POP, NNTP, NetBIOS, SUNRPC, SUNNFS, DCOM, and CUSTOM	Specify a port. You can specify a custom value or a range by choosing **CUSTOM**. If **CUSTOM** is used, Destination Start Port and Destination End Port must be set.
Destination start port	<1-65535>	Specify the start of a port range when Destination Port is set to **CUSTOM**. The value must be less than, or equal to, Destination End Port.
Destination end port	<1-65535>	Specify the end of a port range when Destination Port is set to **CUSTOM**. The value must be greater than, or equal to, Destination Start Port.

Adding an Outbound Filter for a Firewall Filter Interface

To add an outbound filter

1. Click the **Outbound Filter Rules** tab.
   The Outbound Filter Rules details panel appears.
2. Click **Add**.
   The **Add Outbound Filter Rules** dialog box appears.
3. Configure the Outbound Filter Rule settings. These settings are described in the table in Table 159.
4. Click **OK.**

To modify an outbound filter

1. Click the **Outbound Filter Rules** tab.
   The Outbound Filter Rules details panel appears.
2. Select the Outbound Filter you want to modify.
3. Click **Modify**.
   The **Modify Outbound Filter Rule** dialog box appears.
4. Modify the Outbound Filter attributes.
5. Click **OK**.

To delete an outbound filter

1. Click the **Outbound Filter Rules** tab.
   The Outbound Filter Rules details panel appears.
2. Select the Outbound Filter you want to delete.
3. Click **Delete**.
   A message appears that asks you to confirm the deletion.
4. Click **Yes**.

Configuring the order of the inbound and outbound filters for an interface

When you create filters, you assign the order in which the filters are applied.

The order of the filter rules is very important. The more specific rules, such as rules for specific port numbers and addresses, should be placed first. TCP and UDP rules are typically more specific and should be first. Rules for just the IP protocol should be placed last, because they typically ignore port numbers and only match on IP addresses.

The following two examples show how the order of the rules affects what traffic can pass through the IP Firewall when the **Use first match** check box is selected.

Example 1: Rule 1 is configured to Pass TCP protocol 25 from any IP address to 10.10.10.20. Rule 2 is configured to Block any TCP protocol from any IP address to any IP address. If Rule 2 is placed before Rule 1, then Rule 1 will never be reached because all TCP protocol 25 packets destined for IP address 10.10.10.20 will be blocked by Rule 2 first.

Example 2: Rule 1 is configured to Pass TCP protocol 6800 from IP address 192.168.10.20 to IP address 10.10.10.20. Rule 2 is configured to Block all IP protocols from any IP address to any IP address. If Rule 2 is placed before Rule 1, all TCP packets will match Rule 2 first and will be blocked.

To configure the order of the inbound filters

1. Click the **Inbound Filter Rule** tab.
   The Inbound Filter Rule details panel appears.
2. Select the rule that you want to move.
3. Click **Modify**.
   The **Modify Inbound Filter Rule** dialog box appears.
4. Enter the new sequence number.
5. Click **OK**.

To configure the order of the outbound filters

1. Click the **Outbound Filter Rules** tab.
   The Outbound Filter Rules details panel appears.
2. Select the rule that you want to move.
3. Click **Modify**.
   The **Modify Outbound Filter Rule** dialog box appears.
4. Enter the new sequence number.
5. Click **OK**.

Accessing Element Manager through the Firewall

- Do not set any blocking inbound rules on the interface that you use to connect to BCM using Element Manager. This includes enabling the default rules.
- Configure the rule as follows:

Table 160 Inbound Rule Configuration for Element Manager - TCP

Type of filter	Inbound Filter
Protocol	TCP
Source IP Type	Fixed
Source IP	IP address of the system that will access BCM
Source Range Mask	255.255.255.255 (or as appropriate)
Source Port Range	ALL
Destination IP Type	Fixed (or Dynamic if the IP address is remotely assigned)
Destination IP	IP address for this interface (or blank if IP Type is Dynamic)
Destination Range Mask	Appropriate mask (or blank if IP Type is Dynamic)
Destination Port Range	5989-5989

Note: This rule must come before more general rules.

Firewall rules for BCM with Dialup interfaces

For systems with dialup interfaces (ISDN, V.92), we recommend that you add Filters to all interfaces except the dialup interface that blocks NetBIOS traffic. This prevents any NetBIOS packets from getting into the BCM and bringing up the dialup interface link.

Table 161 Inbound Rule Configuration for systems with dialup interfaces

IR1	
Direction:	In
Stateful:	Yes
Disposition:	Block
Protocol:	TCP/UDP
Source IP:	0.0.0.0
Source Mask:	0.0.0.0
Source Port:	NETBIOS
Destination IP:	0.0.0.0
Destination Mask:	0.0.0.0
Destination Port:	NETBIOS

IR2	
Direction:	In
Stateful:	Yes
Disposition:	Block
Protocol:	TCP/UDP
Source IP:	0.0.0.0
Source Mask:	0.0.0.0
Source Port:	NETBIOS
Destination IP:	0.0.0.0
Destination Mask:	0.0.0.0
Destination Port:	DNS

For example, if a BCM is configured with two LANs and one ISDN dialout interface, then these Firewall rules should be placed on both of the LANs.

Chapter 68
Virtual Private Networks (VPN)

BCM uses the Internet and tunneling protocols to create secure extranets. These secure extranets require a protocol for safe transport from the BCM to another device through the Public Data Network (PDN). BCM uses the IPSec ("IPSec" on page 643) tunneling protocol.

Extranets can connect:

- mobile users to a fixed private network at their office over the PDN
- private networks in the two branch offices of the same corporation over PDN
- two divisions of the same corporation over the corporate intranet

When connecting two branch offices, the use of a VPN over the public data network is very efficient if the connection is required only intermittently or a dedicated point-to-point link is considered too expensive. Also, with the advent of business-to-business solutions, VPNs can be deployed to provide secure connections between corporations.

IPSec tunnel modes

In the IPSec Specification, there are two tunnel modes defined: tunnel mode and transport mode. BCM supports only tunnel mode. Tunnel mode describes a method of packetizing TCP/IP traffic to create a virtual tunnel.

Tunnels are created between servers, which are also known as gateways. This is called a Branch Office Connection. The end nodes connect to each other through gateways. These gateways set up the tunnel over the PDN on behalf of the end nodes. The establishment of the tunnel, and the PDN in between, is transparent to the end nodes which behave as if they are interacting through a router. Typically, the edge devices connecting the branches of a corporation to the ISP use VPN in this mode.

BCM is compatible with the Extranet Switch and the Nortel Services Edge Router (formerly known as Shasta 5000).

Refer to the following topics to configure the tunnel portion of BCM using IPSec.

IP Addresses and DHCP Server

Ensure that the IP addresses for the LAN interfaces, WAN interfaces, and dial up links are unique across all sites. This simplifies configuration, eliminates conflicts due to NAT, and prevents the addresses assigned by the DHCP server from conflicting with the IP addresses of subnets in remote sites.

For information about how to change the DHCP Server settings, refer to the "Configuring DHCP" on page 583.

DNS Server

The following configuration is recommended if you are using a DNS Server:

- Choose one of the offices to act as the primary office. The server in primary office must have a dedicated link to the Internet.
- Make the server in the primary office the primary domain server. Ensure the DNS Server in the primary office contains all of the entries for allow the branch offices.
- Configure the DNS Servers in the branch offices to run in cache mode only.
- Configure the branch DNS servers to forward DNS Server requests to the Internet Service Provider first and then to the DNS Server in the primary office.

For information about how to configure the DNS proxy service on BCM, refer to the “Configuring DNS” on page 679.

IPSec

The IPsec tunneling protocol is supported by Nortel and other third-party vendors. IPsec is a standard that offers a strong level of encryption (DES, Triple DES and AES), integrity protection (MD5 and SHA), and the IETF-recommended Internet Security Association & Key Management Protocol (ISAKMP) and Oakley Key Determination Protocols.

Refer to the following topics for additional information:

- "Encryption" on page 644
- "Settings required for IPSec tunnels" on page 646
- "IPSec Branch Office Tunnel configuration" on page 656
- "Creating a tunnel between two BCMs" on page 665
- "IPSec Remote User Tunnel configuration" on page 666
- "IPSec Remote User Tunnel configuration" on page 666
- "Adding a Remote User IPSec Tunnel" on page 669

IPsec offers the following features

- Branch Office support that allows you to configure an IPSec tunnel connection between two private networks.
- Client support is via the Contivity VPN client. The BCM supports VPN client support from a remote computer with version of the VPN Client installed. No special ISP services are required.
- Support for IP address translation via encapsulation, packet-by-packet authentication.
- Strong encryption and token codes.

Encryption

All of the following encryption methods ensure that the packets have come from the original source at the secure end of the tunnel.

Table 162 shows a comparison of the security provided by the available encryption and authentication methods.

Table 162 Comparing Encryption and Authentication Methods

Method (strongest to weakest)	Encryption of IP Packet Payload	Authentication of IP Packet Payload	Authentication of Entire IP Packet
ESP-AES128-SHA1	Yes	Yes	No
ESP-3DES-SHA1	Yes	Yes	No
ESP-3DES-MD5	Yes	Yes	No
ESP-DES56-SHA1	Yes	Yes	No
ESP-DES56-MD5	Yes	Yes	No
AH HMAC SHA1	No	No	Yes
AH HMAC MD5	No	No	Yes

➡ **Note:** Using higher-level encryption, such as AES-128, requires more system resources and increases packet latency. You need to consider this when designing your overall network.

➡ **Note:** If two devices have different encryption settings, the two devices will negotiate downward until they agree on a compatible encryption capability. For example, if Switch A attempts to negotiate Triple DES encryption with Switch B that is using 56-bit DES, then the Switch B will reject Triple DES encryption in favor of the 56-bit DES.

Each of the systems must have at least one encryption setting in common. If they do not, a tunnel will not be negotiated. In the example above, both systems must have 56-bit DES enabled.

The encryption level you choose is made of three components:

- the protocol
- the encryption method
- the authentication method

Protocol

The protocol can be ESP or AH.

- ESP
  Encapsulating Security Payload (ESP) provides data integrity, source authentication and confidentiality for IP datagrams by encrypting the payload data to be protected. ESP uses the Data Encryption Standard (DES) and Triple DES and AES algorithms.
- AH
  Authentication Header (AH) provides data integrity and source authentication. The AH method does not encrypt data.

Note: The use of a NAT device in the IPSec tunnel path can sometimes cause the AH method to report a security violation. This occurs because the NAT device changes the IP Address of an AH authenticated packet causing the authentication of this packet to fail.

Encryption method

The encryption method can be AES, Triple DES, 56-bit DES. AES is the strongest encryption and 56-bit DES is the weakest encryption.

- AES
  Advanced Encryption Standard (AES) is a symmetric key encryption technique. The AES algorithm uses one of three cipher key strengths: a 128-, 192-, or 256-bit encryption key. BCM 4.0 supports 128 and 256 bit key strengths.
- Triple DES
  Triple DES is an encryption block cipher algorithm that uses a 168-bit key. It uses the DES encryption algorithm three times. The first 56 bits of the key is used to encrypt the data, then the second 56 bits is used to decrypt the data. Finally, the data is encrypted once again with the third 56 bits. These three steps triple the complexity of the algorithm.
- 56-bit DES
  56-bit DES is an encryption block cipher algorithm that uses a 56-bit key (with 8 bits of parity) over a 64-bit block. The 56 bits of the key are transformed and combined with a 64-bit message through a complex process of 16 steps.

Authentication method

The authentication method can be SHA1 or MD5.

- SHA1
  Secure Hash Algorithm (SHA1) produces a 160-bit hash. It is regarded by cryptographers as being more resistant to attacks than MD5. SHA1 does not encrypt data.
- MD5
  Message Digest 5 (MD5) Algorithm produces a 128-bit hash. It is used to confirm the authenticity of a packet. MD5 does not encrypt data. Also, MD5 provides integrity that detects packet modifications.

Both SHA1 and MD5 use Hashed Message Authentication Code (HMAC) to improve authentication. HMAC is a technique that uses a secret key and a message digest function to create a secret message authentication code.

IPSec capacity restrictions

The BCM performs all IPSec processing using software. To prevent overloading the BCM processor with IPSec traffic processing, the network traffic that requires IPSec processing should not exceed 6Mbps. This is based on using 3DES encryption with SHA authentication.

Note: The maximum number of concurrent tunnels the BCM supports is 16. However, this number could be less depending on the configuration.

Consider the following factors when determining maximum IPSec capacity:

- Tunnel negotiation
  Since tunnel negotiation requires a significant amount of processing time, the number of tunnels that are negotiated at one time should be limited. The tunnels are re-negotiated based on either the Rekey Timeout or the Rekey Data Count. If a number of tunnels will be running concurrently, you should stagger these values.

Settings required for IPSec tunnels

The data packets that pass through IPSec tunnels interact with other routing features in BCM. As a result, there are several settings you must make in other features for IPSec tunnels to operate.

NAT (Network Address Translation)

Note: In this section, the term LAN is referring to the Local Accessible Network.

BCM does not support NAT on the Local Endpoint of an IPSec Tunnel.

Packets can be sent through an IPSec tunnel with or without NAT applied. To send packets through the tunnel with NAT applied, configure the LAN to include only a network for the endpoint itself. For example, if the Local Endpoint is 10.10.13.2, then the LAN would be 10.10.13.2 with a mask of 255.255.255.255. To send packets through the tunnel without NAT applied, configure the LANs with the local Private IP network(s) and the Remote Accessible Networks with the networks on the other side of the Remote Endpoint. Using the above example, we know that the other interfaces on the local BCM have IP addresses of 10.10.10.1 and 10.10.11.1. The remote BCM has a subnet of 12.12.12.1. Therefore, the LAN would have two networks configured as 10.10.10.0 with a mask 255.255.255.0 and 10.10.11.0 with a mask 255.255.255.0 and the Remote Accessible Networks would be 12.12.12.0 with a mask of 255.255.255.0. All packets that do not match these rules will be NATed and sent out the interface and not through the tunnel. This is a useful configuration if access to both the Internet and the other side of an IPSec tunnel is desired.

Dialup ISDN connections

When you are creating an IPSec tunnel over a Dialup ISDN connection, the endpoint must have a fixed IP address.

Compatibility with Extranet Switch and Nortel Services Edge Router 5500 (formerly known as Shasta 5000)

When connecting to a Contivity Server, you must disable Vendor ID, Nailed Up and Compression on the Contivity Server.

BCM does not support the IPSec RIP implementation used by the Contivity Server. Use Static Routes when connecting to the Extranet Switch.

When connecting to a Nortel Services Edge Router (formerly known as Shasta 5000), you must disable Perfect Forward Secrecy (PFS) on the BCM Branch Office Tunnel Configuration.

Multiple IP Address restrictions

Although the BCM supports the configuration of additional IP addresses on its network interfaces, IPSec does not currently support the use of these additional IP addresses for Branch Office Local Endpoint Addresses, Remote Endpoint Addresses or the Destination IP Address for IPSec VPN Clients.

Firewall rules for IPSec Branch Office and Remote User Tunnels

In order to allow IPSec packets through the firewall interface which blocks all incoming packets, a number of rules must be configured. In addition to allowing the IPSec packets through, you must also remember to create rules to allow the packets that come through the tunnel.

Firewall Rules for Branch Office

In the Branch office case, four rules must be created. One is for the key exchange protocol (IKE), the other two are for the type of protocol used (ESP and/or AH). The fourth rule is for the inbound rules. Table 163, Table 164, Table 165 and Table 166 show the rules required.

Table 163 Rule 1

Protocol	UDP
Source IP	Remote Endpoint address of Branch Office Tunnel
Source Mask	255.255.255.255
Source Port	500
Destination IP	Local Endpoint address of Branch Office Tunnel
Destination Mask	255.255.255.255
Destination Port	500

Table 164 Rule 2

Protocol	IPSEC_ESP
Source IP	Remote Endpoint address of Branch Office Tunnel
Source Mask	255.255.255.255

Table 164 Rule 2

Destination IP	Local Endpoint address of Branch Office Tunnel
Destination Mask	255.255.255.255

Table 165 Rule 3

Protocol	**IPSEC_AH**
Source IP	Remote Endpoint address of Branch Office Tunnel
Source Mask	255.255.255.255
Destination IP	Local Endpoint address of Branch Office Tunnel
Destination Mask	255.255.255.255

Table 166 Rule 4

Protocol	**IGNORE**
Source IP	Remote Accessible Network of Branch Office Tunnel
Source Mask	Remote Accessible Network Subnet Mask of Branch Office Tunnel
Destination IP	Local Accessible Network of Branch Office Tunnel
Destination Mask	Local Accessible Network Subnet Mask of Branch Office Tunnel

Example of Rules

Figure 200 Branch office example

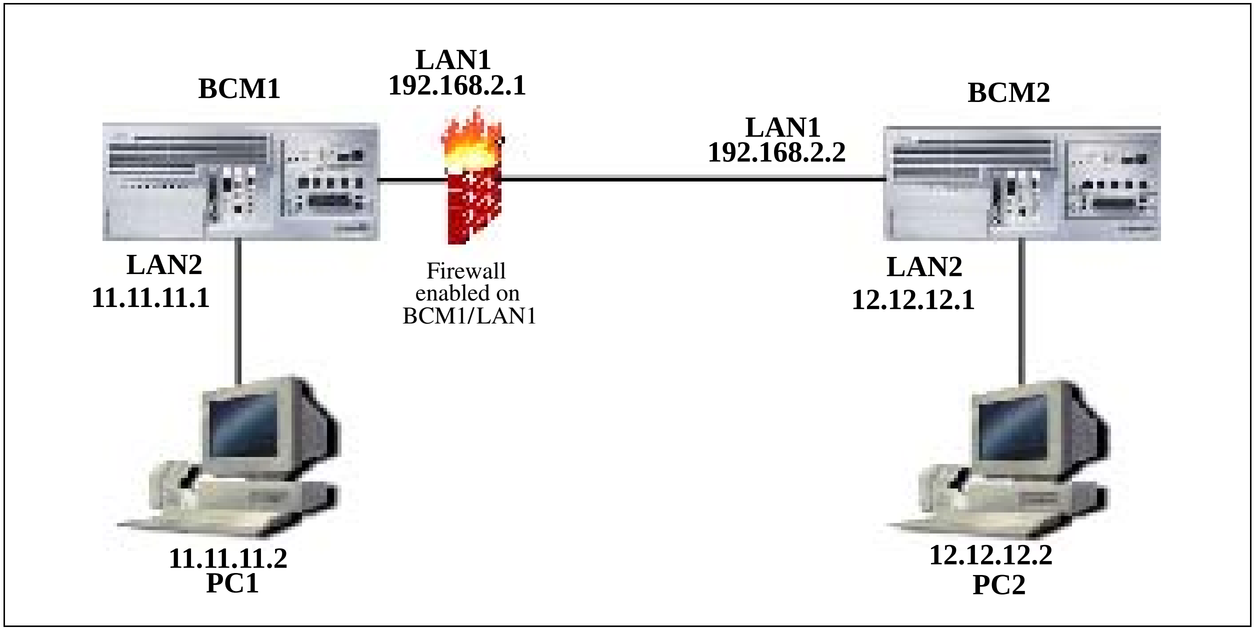


Table 167 Rule 1

Protocol	**UDP**
Source IP	192.168.2.2
Source Mask	255.255.255.255
Source Port	500
Destination IP	192.168.2.1
Destination Mask	255.255.255.255
Destination Port	500

Table 168 Rule 2

Protocol	IPSEC_ESP
Source IP	192.168.2.2
Source Mask	255.255.255.255
Destination IP	192.168.2.1
Destination Mask	255.255.255.255

Table 169 Rule 3

Protocol	IPSEC_AH
Source IP	192.168.2.2
Source Mask	255.255.255.255
Destination IP	192.168.2.1
Destination Mask	255.255.255.255

Table 170 Rule 4

Protocol	IGNORE
Source IP	12.12.12.0
Source Mask	255.255.255.0
Destination IP	11.11.11.0
Destination Mask	255.255.255.0

Firewall rules for Remote User Tunnels

In the Remote User Tunnel case, four rules must be created. One is for the key exchange protocol (IKE), the other two are for the type of protocol used (ESP and/or AH). The fourth rule is for the inbound rules. Table 171, Table 172, Table 173 and Table 174 show the rules required.

Table 171 Rule 1

Protocol	UDP
Source IP	IP Address of client PC (or 0.0.0.0 if not known)
Source Mask	255.255.255.255 (or 0.0.0.0 if not known)
Source Port	500
Destination IP	IP Address of Interface that will receive VPN Client Connection request for client PC
Destination Mask	255.255.255.255
Destination Port	500

Table 172 Rule 2

Protocol	IPSEC_ESP
Source IP	IP Address of client PC (or 0.0.0.0 if not known)
Source Mask	255.255.255.255 (or 0.0.0.0 if not known)
Destination IP	IP Address of Interface that will receive VPN Client Connection request for client PC
Destination Mask	255.255.255.255

Table 173 Rule 3

Protocol	IPSEC_AH
Source IP	IP Address of client PC (or 0.0.0.0 if not known)
Source Mask	255.255.255.255 (or 0.0.0.0 if not known)
Destination IP	IP Address of Interface that will receive VPN Client Connection request for client PC.
Destination mask	255.255.255.255

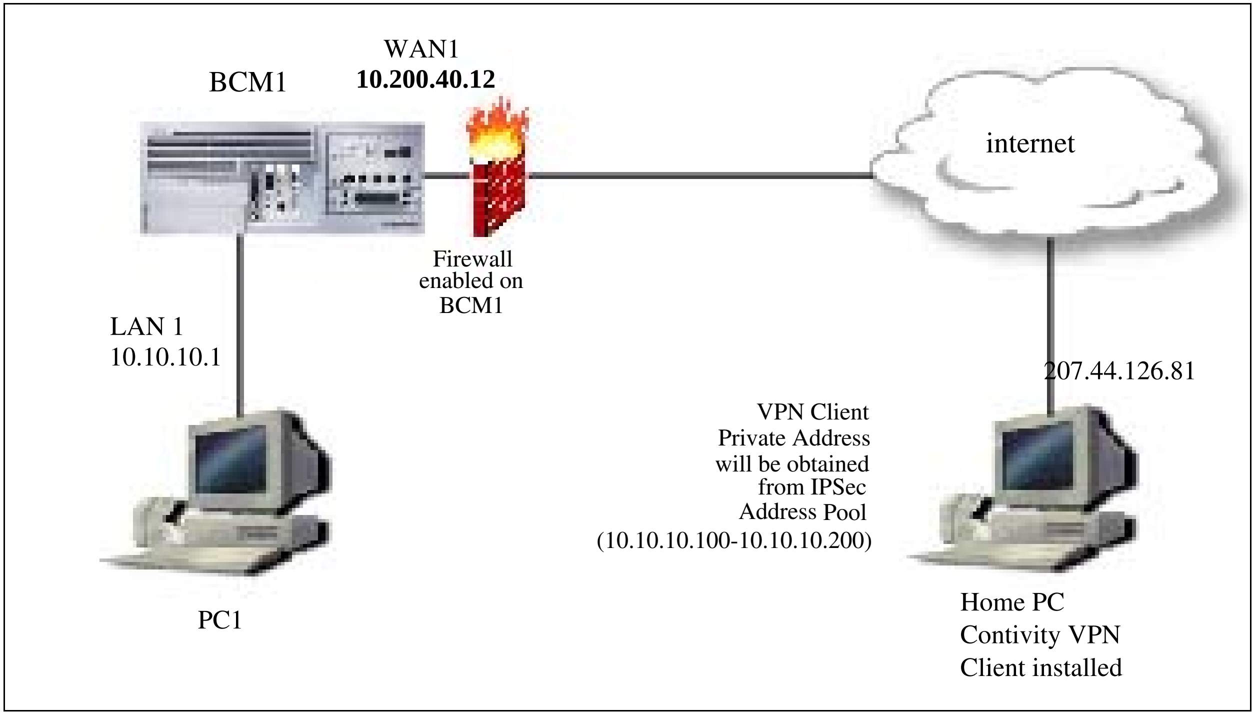
Table 174 Rule 4

Protocol	IGNORE
Source IP	IP Address from private network assigned to VPN client (or network address if IP Address Pool used).
Source Mask	255.255.255.255 (or Subnet Mask assigned if IP Address Pool used)
Destination IP	IP Network Address of Private network
Destination Mask	Subnet Mask of Private network

Example 2

BCM 1 has been configured with a WAN1 address of 10.200.40.12 and a LAN1 address of 10.10.10.1. Your computer at home has the address of 207.44.126.81. An IPSec Remote User tunnel has been configured on BCM1. This tunnel will obtain it’s IP Address from an IPSec Address Pool. This IPSec Address Pool is configured with range: 10.10.10.100 - 10.10.10.200 with a subnet mask of 255.255.255.0. You only allow ESP as the IPSec protocol. Firewall is enabled on LAN1.You will need the following rules:

Figure 201 Remote User Tunnel example


BCM1
WAN1
10.200.40.12
internet
Firewall enabled on BCM1
LAN 1
10.10.10.1
207.44.126.81
VPN Client Private Address will be obtained from IPSec Address Pool (10.10.10.100-10.10.10.200)
PC1
Home PC Contivity VPN Client installed

Rule 1	
Protocol	UDP
Source IP	207.44.126.81
Source Mask	255.255.255.255
Source Port	500
Destination IP	10.200.40.12
Destination Mask	255.255.255.255
Destination Port	500

Rule 2	
Protocol	IPSEC_ESP
Source IP	207.44.126.81
Source Mask	255.255.255.255
Destination IP	10.200.40.12
Destination Mask	255.255.255.255

Note: Rules 3 is technically not required as only ESP is being used.

Rule 3	
Protocol	IPSEC_AH
Source IP	207.44.126.81
Source Mask	255.255.255.255
Destination IP	10.200.40.12
Destination Mask	255.255.255.255

Rule 4	
Protocol	IGNORE
Source IP	10.10.10.0
Source Mask	255.255.255.0
Destination IP	10.10.10.0
Destination Mask	255.255.255.0

For information about how to add or change Filters, refer to "Configuring IP Filter Rules" on page 615.

Creating Firewall Rules Automatically

You can create these rules automatically when creating or modifying Branch Office and Remote User Tunnels by selecting the Create Firewall Rules on the Parameters page for a particular tunnel. The four firewall rules required by the Branch Office tunnels are then created. You can view these rules on the Input Filters' Rule Setting panel for the interface used. If the Branch Office tunnel is enabled and IPSec is enabled globally, then the four rules created are added to the front of the Rule Order that appears on the Input Filters' Rule Order panel for the interface used. If the Branch Office tunnel is later disabled, then the rules are removed from the Rule Order, but still exist on the Input Filters' Rule Setting panel. If the user does not select the Create Firewall Rules option, then the three firewall rules created for the Branch Office tunnel are deleted.

You can also create firewall rules for Remote User tunnels. The rule creation process is the same as for Branch Office tunnels except that the user must select which interface they want to create firewall rules for. The four rules in Table 163 to Table 166 are created for Remote User Tunnels if you select the check box for the Create Firewall Rules option.

Changing the IPSec global settings

The IPSec global settings apply to all of the IPSec tunnels.

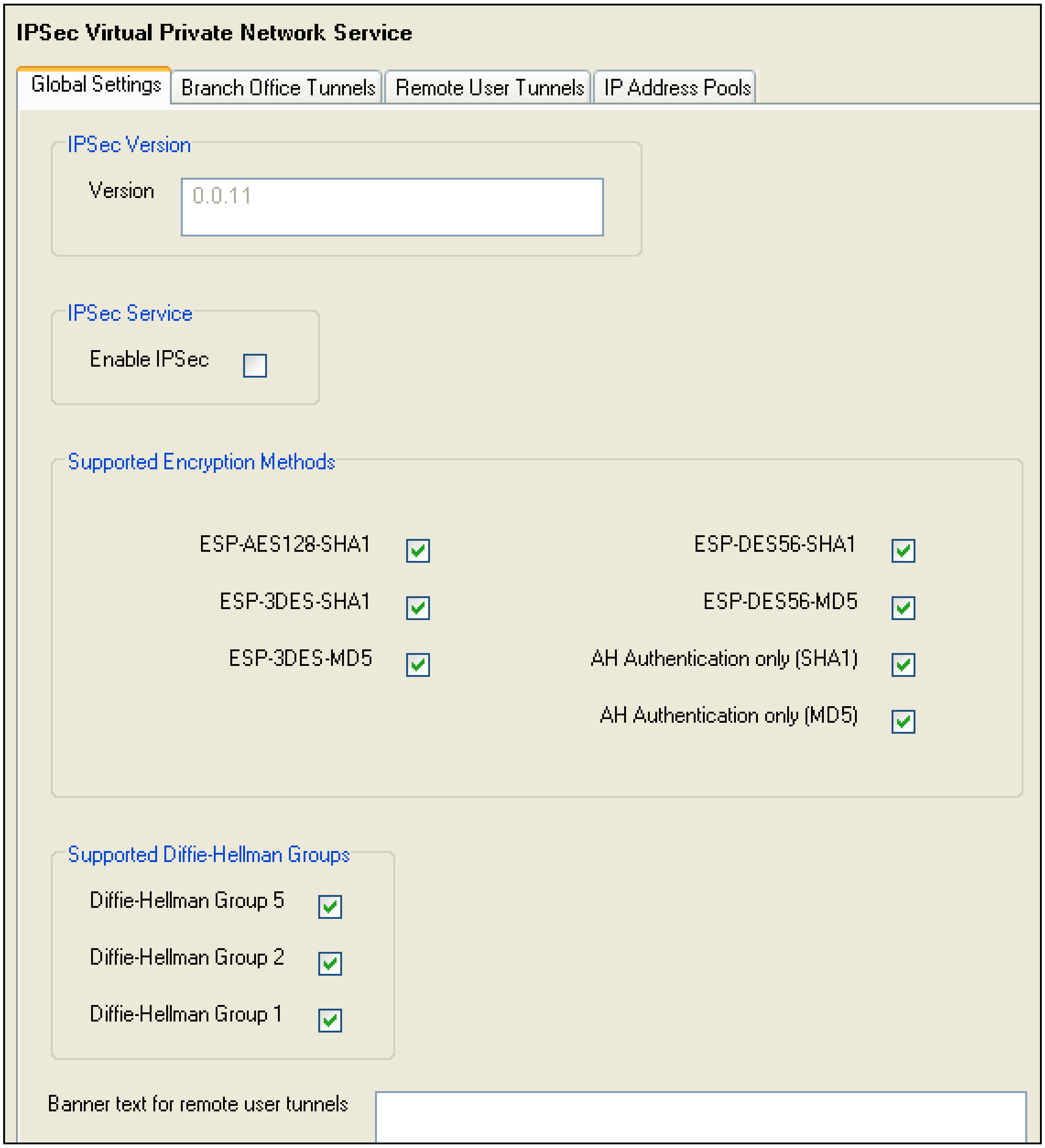
To change the IPSec global settings

1 Click **Configuration > Data Services > IPsec VPN**.

2 The **Global Settings** panel appears. See Figure 202.

3 Configure the IPSec global settings. Refer to the information in Table 25.

Figure 202 Global Settings Panel



IPSec Virtual Private Network Service

IPSec Version

Version: 0.0.11

IPSec Service

Enable IPSec □

Supported Encryption Methods

ESP-AES128-SHA1 ☑
ESP-3DES-SHA1 ☑
ESP-3DES-MD5 ☑
ESP-DES56-SHA1 ☑
ESP-DES56-MD5 ☑
AH Authentication only (SHA1) ☑
AH Authentication only (MD5) ☑

Supported Diffie-Hellman Groups

Diffie-Hellman Group 5 ☑
Diffie-Hellman Group 2 ☑
Diffie-Hellman Group 1 ☑

Banner text for remote user tunnels

Table 25 IPSec Global settings (Sheet 1 of 2)

Attribute	Value	Description
IPSec Version		
Version	<read-only>	Displays the version number of the IPSec service.
IPSec Service		
Enable IPSec	<check box>	Select to enable IPSec Tunnel

Table 25 IPSec Global settings (Sheet 2 of 2)

Attribute	Value	Description
Supported Encryption Methods		
Encryption	ESP-AES128-SHA1 ESP-3DES-SHA1 ESP-3DES-MD5 ESP-DES56-SHA1 ESP-DES56-MD5 AH Authentication only (SHA1) AH Authentication only (MD5)	Select the encryption levels that you allow your IPSec tunnels to use. The encryption level used for the IPSec tunnel is negotiated when the tunnel is opened. The encryption levels you select are the encryption levels that you allow BCM to use for IPSec tunnels. This is a global setting that applies to all of the IPSec tunnels on BCM. When you add an IPSec tunnel, you can further restrict the encryption levels for each tunnel. For more information, refer to "Branch Office IPSec Tunnel" on page 656. For a description of the encryption levels, refer to "Encryption" on page 644.
Supported Diffie-Hellman Groups		
Protocol	Diffie-Hellman Group 5 (1536-bit) Diffie-Hellman Group 2 (1024-bit) Diffie-Hellman Group 1 (768-bit) Default is Group 2	Diffie-Hellman is a public-key cryptographic protocol that allows two parties to establish a shared secret over an insecure communications channel. It is also used within IKE (Internet Key Exchange) to establish session keys. IPSec uses the Diffie-Hellman algorithm to provide the keying material for all other encryption keys. Higher (larger bit keying material) Diffie-Hellman groups provide more security but require more processor time.
Banner text for remote user tunnels	<alphanumeric>	Banner Text is the text that appears when a remote user logs into the BCM using the IPSec VPN Client. You can use this text to display important information (such as security information) to the remote user. You can enter a maximum 1000 ASCII characters

IPSec Branch Office Tunnel configuration

The branch office feature allows you to configure an IPsec tunnel connection between two private networks. Typically, one private network is behind a locally configured switch while the other is behind a remote switch. A branch office configuration allows you to configure the accessible subnetworks behind each switch. The configuration also contains the information that is necessary to set up the connection, such as the switch IP addresses, encryption types and authentication methods.

Refer to the following procedures:

- "To add a branch office IPSec tunnel" on page 657
- "To modify a Branch Office IPSec Tunnel" on page 662
- "To delete a branch office IPSec tunnel" on page 664

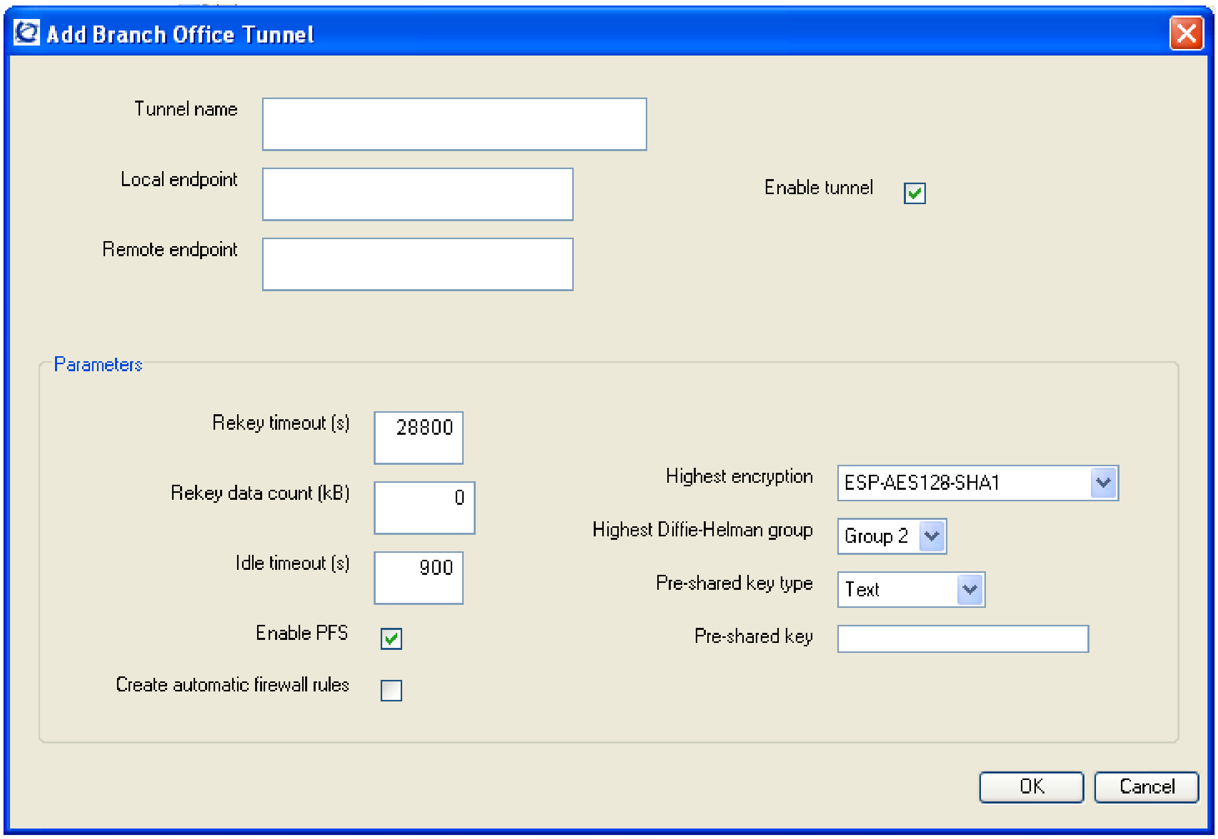
Branch Office IPSec Tunnel

A Branch Office IPSec Tunnel connects two offices together. The IPSec Tunnel connects the local Business Communications Manager system to another Business Communications Manager system, an Extranet Switch or a Nortel Services Edge Router (formerly known as Shasta 5000) switch.

To add a branch office IPSec tunnel

1. Click **Configuration > Data Services > IPSec VPN**.
2. Click **Branch Office Tunnels** tab.
   The Branch Office Tunnels panel appears.
3. Click **Add**.
   The **Add Branch Office Tunnel** panel appears. See Figure 203
4. Configure the Branch Office Tunnel Settings. Refer to the information in Table 26.
5. Click **OK**.

Figure 203 Branch Office Tunnel



Tunnel name

Local endpoint

Remote endpoint

Enable tunnel ☑

Parameters

Rekey timeout (s) 28800

Rekey data count (kB) 0

Idle timeout (s) 900

Enable PFS ☑

Create automatic firewall rules □

Highest encryption ESP-AES128-SHA1

Highest Diffie-Helman group Group 2

Pre-shared key type Text

Pre-shared key

Table 26 IPSec Branch Office Tunnel settings (Sheet 1 of 3)

Attribute	Value	Description
Tunnel name	<alphanumeric>	Specify the Tunnel identifier. The Tunnel Name uniquely identifies a IPSec tunnel. The value for this setting must follow certain conventions. You must type the prefix 'T' followed by a unique number identifying the IPSec Tunnel. For example, 'T2' is a valid name. If you specify an existing Tunnel number, you receive an error message. The Tunnel identifier does not have any significance, other than uniquely identifying an entry. **Note**: The maximum number of tunnels you can add is 20.
Local endpoint	<IP address>	Specify the IP address of the interface on BCM that is the entrance or exit of the IPSec tunnel.
Remote endpoint	<IP address>	Specify the IP address of the remote that is the entrance or exit of the IPSec tunnel. **Note**: Different tunnels cannot have the same Remote Endpoint.
Enable tunnel	<check box>	Select to enable IPSec Tunnel.
Parameters		

Table 26 IPSec Branch Office Tunnel settings (Sheet 2 of 3)

Attribute	Value	Description
Rekey timeout (s)	<0-359999>	Specify the amount of time you can use a key before the tunnel is re-negotiated. You should limit the lifetime of a single key used to encrypt data or else you will compromise the effectiveness of a single session key. Use the Rekey Timeout setting to control how often new session keys are exchanged between servers. You cannot set the Rekey Timeout setting to less than three minutes, except to disable the timeout by entering 0. **Note:** A setting of 0 disables the Rekey Timeout setting. Default: 28800 secs
Rekey data count (kB)	<0 -1000000>	Specify the amount of data you can transmit on the tunnel before the tunnel is re-negotiated. A setting of 0 disables the Rekey Data Count. **Note**: If you set the Rekey Data Count too low, the tunnel is re-negotiated too often and will consume extra system resources. Default: 0 Kbytes
Idle timeout (s)	<0-359999>	Specify the amount of time the tunnel can remain idle before the tunnel is closed. You cannot set the Idle Timeout setting to less than three minutes, except to disable the timeout by entering 0. **Note**: A setting of 0 disables the Idle Timeout setting. Default: 900 secs
Enable PFS	<check box>	Enable Perfect Forward Secrecy (PFS). With PFS, keys are not derived from previous keys. This ensures that one key being compromised cannot result in the compromise of subsequent keys. If you create a tunnel to a Contivity Extranet Switch, you must select Enable PFS. **Note**: Clear Enable PFS for connections to the Nortel Services Edge Router (formerly known as Shasta 5000).
Create automatic firewall rules	<check box>	Select this check box if you want the BCM to create Firewall rules that allow traffic for this tunnel to pass through the Firewall. Clear this check box if you do not want BCM to create Firewall rules for this tunnel. If you are using the BCM Firewall, Nortel recommends that you select this option. See Figure 200 for an example of a branch office firewall configuration.
Highest encryption	<drop-down list>	Select the highest encryption level allowed on this IPSec tunnel. When the encryption level is negotiated for this tunnel, BCM will not use any encryption level higher than the encryption level specified in this field. For a description of the encryption levels, refer to “Encryption” on page 644.
Highest Diffie-Hellman Group	Diffie-Hellman Group 5 (1536-bit) Diffie-Hellman Group 2 (1024-bit) Diffie-Hellman Group 1 (768-bit)	Diffie-Hellman is a public-key cryptographic protocol that allows two parties to establish a shared secret over an insecure communications channel. It is also used within IKE (Internet Key Exchange) to establish session keys. IPSec uses the Diffie-Hellman algorithm to provide the keying material for all other encryption keys. Higher (larger bit keying material) Diffie-Hellman groups provide more security but require more processor time. Default: Group 2

Table 26 IPSec Branch Office Tunnel settings (Sheet 3 of 3)

Attribute	Value	Description
Pre-shared key type	Text Hexadecimal	Select the format for the Pre-shared key. The Key Type must be the same on both ends of the IPSec tunnel. The format can be text or hexadecimal. **Note**: If you change the Key Type, the Pre-shared key is deleted.
Pre-shared key	<alphanumeric>	Specify the text or hexadecimal string used to authenticate the data sent on this tunnel. The maximum length of the Pre-shared Key is 32 characters. This key must be used at both ends of the IPSec Tunnel. For best security, use a secure method to share this key.
Actions		
OK	<button>	Click to add Branch office tunnel.
Cancel	<button>	Click to cancel operation.

Local Accessible Networks to the Branch Office IPSec tunnel.

Note: The maximum number of Local Accessible Networks you can add is 16.

To add a Local Accessible Network to the Branch Office IPSec tunnel

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Branch Office Tunnels** tab.

3 Click the tunnel you want to modify.
The details panel appears.

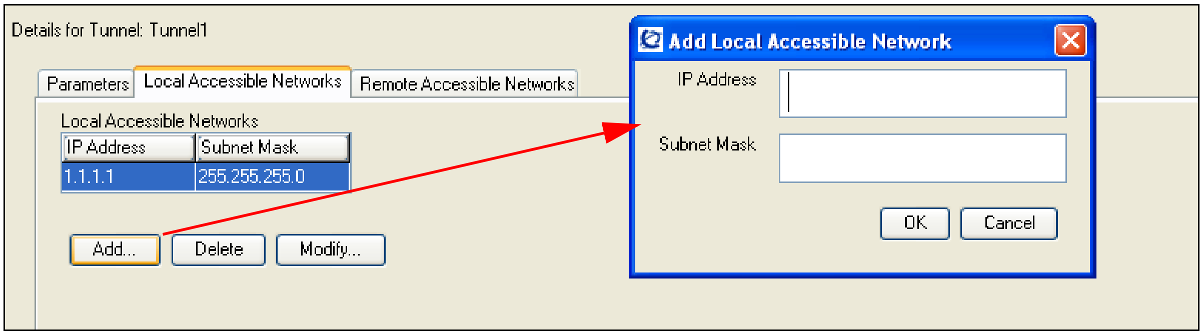
4 Click the **Local Accessible Networks** tab.
The Local Accessible Networks panel appears. See Figure 204.

5 Click **Add**.

6 Configure the Local Accessible Network parameters. Refer to the information in Table 175.

7 Click **OK**.

Figure 204 Local Accessible Networks tab



Details for Tunnel: Tunnel1

IP Address	Subnet Mask
1.1.1.1	255.255.255.0

Parameters
Local Accessible Networks
Remote Accessible Networks
Local Accessible Networks
Add...
Delete
Modify...
Add Local Accessible Network
IP Address
Subnet Mask
OK
Cancel

Table 175 IPSec Local Accessible Networks parameters

Attribute	Value	Description
IP Address	<IP address>	Specify the IP addresses of interfaces on BCM that can connect to this tunnel.
Subnet Mask	<IP address>	Specify the subnet mask of interfaces on BCM that can connect to this tunnel.

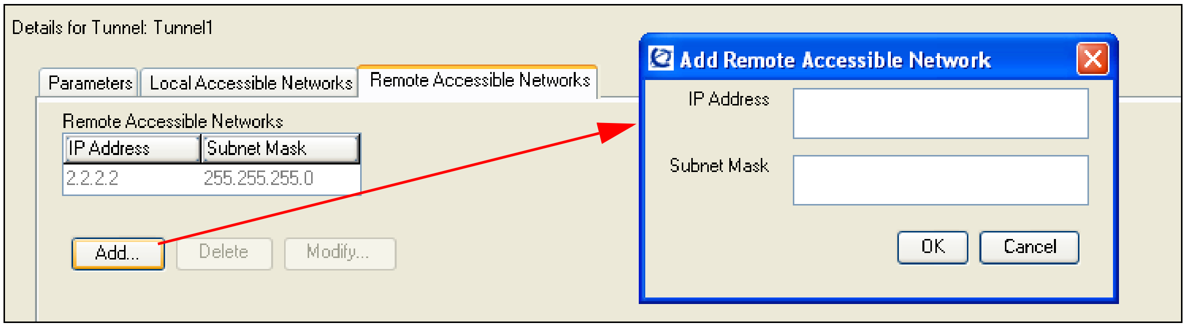
Remote Accessible Networks to the Branch Office IPSec tunnel

Note: The maximum number of Remote Accessible Networks you can add is 16.

To add a Remote Accessible Network to the Branch Office IPSec tunnel

1. Click **Configuration > Data Services > IPSec VPN**.
2. Click the **Branch Office Tunnels** tab.
3. Click the tunnel you want to modify.
   The details panel appears.
4. Click the **Remote Accessible Networks** tab.
   The Remote Accessible Networks Panel appears. See Figure 205.
5. Click the **Add** button.
6. Configure the Remote Accessible Network parameters. Refer to the information in Table 176.
7. Click **OK**.

Figure 205 Remote Accessible Networks tab


Details for Tunnel: Tunnel1

Parameters
Local Accessible Networks
Remote Accessible Networks

Remote Accessible Networks

IP Address	Subnet Mask
2.2.2.2	255.255.255.0

Add...
Delete
Modify...
Add Remote Accessible Network

IP Address

Subnet Mask

Table 176 IPSec Remote Accessible Network parameters

Attribute	Attribute	Description
IP Address	<IP address>	Specify the P addresses of the Remote Accessible Network that you can connect to using this tunnel.
Subnet Mask	<IP address>	Specify the subnet mask of Remote Accessible Network that you can connect to using this tunnel.

Note: Different tunnels cannot have the same Remote Accessible Networks.

Modifying a Branch Office IPSec Tunnel

To modify a Branch Office IPSec Tunnel

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Branch Office Tunnels** tab.

3 Click the tunnel you want to modify.
The details panel appears.

4 Click **Modify**.
The Modify Branch Office Tunnel panel appears.

5 Change the required IPSec Tunnel settings.
For information about the settings refer to "Branch Office IPSec Tunnel" on page 656.

6 Click **OK**.

Modifying Local Accessible Networks to the Branch Office IPSec tunnel

To modify a Local Accessible Network to the Branch Office IPSec tunnel

1. Click **Configuration > Data Services > IPSec VPN**.
2. Click the **Branch Office Tunnels** tab.
3. Click the tunnel you want to modify.
   The details panel appears.
4. Click the **Local Accessible Networks** tab.
   The Local Accessible Networks panel appears.
5. Click the Local Accessible Network you want to modify.
6. Click **Modify**.
   The Local Accessible Networks panel appears. See Figure 204.
7. Modify the Local Accessible Network parameters.
8. Click **OK**.

Modifying Remote Accessible Networks to the Branch Office IPSec tunnel

To modify a Remote Accessible Network to the Branch Office IPSec tunnel

1. Click **Configuration > Data Services > IPSec VPN**.
2. Click the **Branch Office Tunnels** tab.
3. Click the tunnel you want to modify.
   The details panel appears.
4. Click the **Remote Accessible Networks** tab.
   The Remote Accessible Networks panel appears.
5. Click the Remote Accessible Network you want to modify.
6. Click **Modify**.
   The Remote Accessible Networks panel appears. See Figure 205.
7. Modify the Remote Accessible Network parameters.
8. Click **OK**.

Deleting a Branch Office IPSec tunnel

To delete a branch office IPSec tunnel

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Branch Office Tunnels** tab.

3 Click the tunnel you want to delete.

4 Click **Delete**.
A message prompts you to confirm the deletion. See Figure 203.

5 Click **Yes**.

Deleting Local Accessible Networks to the Branch Office IPSec tunnel

To delete Local Accessible Networks to the Branch Office IPSec tunnel

1 Click **Configuration > Data Services > IPSec VPN**

2 Click the **Branch Office Tunnels** tab.

3 Click the tunnel you want to modify.
The details panel appears.

4 Click the **Local Accessible Networks** tab.
The Local Accessible Networks panel appears.

5 Click the Local Accessible Network you want to delete.

6 Click **Delete**.
A message prompts you to confirm the deletion. See Figure 204.

7 Click **Yes**.

Deleting Remote Accessible Networks to the Branch Office IPSec tunnel

To delete a Remote Accessible Networks to the Branch Office IPSec tunnel

1 Click **Configuration > Data Services > IPSec VPN.**

2 Click the **Branch Office Tunnels** tab.

3 Click the tunnel you want to modify.
The details panel appears.

4 Click the **Remote Accessible Networks** tab.
The Remote Accessible Networks panel appears.

5 Click the Remote Accessible Network you want to delete.

6 Click **Delete**.
A message prompts you to confirm the deletion. See Figure 205.

7 Click **Yes**.

Creating a tunnel between two BCMs

The following is an example of a how to connect two BCM systems together using an IPSec tunnel.

In this example, the IPSec tunnel connects a BCM with a LAN 2 IP address of 10.10.11.1 and another BCM with a LAN 2 IP address of 10.10.11.2. LAN 1 on the first BCM is on the subnet 12.12.12.0. The LAN 1 of the second BCM is on subnet 14.14.14.0.

Configuring the first BCM

1 Click **Configuration > Data Services > IPSec VPN**

2 Click the **Global Settings** tab.

3 Select the **Enable IPSec** check box.

4 Click the **Branch Office Tunnels** tab.

5 Click **Add**.
The Add Branch Office Tunnel panel appears.

6 Set the Tunnel name to Example.

7 Set the Local Endpoint to **10.10.11.1**.

8 Set the Remote Endpoint to **10.10.11.2**.

9 Set the Pre-shared key type to **Text**.

10 Set the Pre-shared key to **123**.

11 Click **OK**.
A confirmation box will appear prompting you to re-enter the Pre-shared key
Re-enter your Pre-shared key and click **OK** to save.

12 Click **OK** on the Add Branch Office Tunnel panel

13 Select the Branch Office Tunnel you just created.
The details panel appears.

14 Click the **Local Accessible Networks** tab.

15 Click **Add**.
The Add Local Accessible Network panel appears.

16 Set the IP Address to **12.12.12.0** with Subnet mask **255.255.255.0**.
Click **OK.**

17 Click the **Remote Accessible Networks** tab.

18 Click **Add**.
The Add Remote Accessible Network panel appears.

19 Set the IP Address to **14.14.14.0** with Subnet mask **255.255.255.0**.

20 Click **OK**.

Configuring the second BCM

1 Click **Configuration > Data Services > IPSec VPN.**

2 Click the **Branch Office Tunnels** tab.

3 Click **Add**.
The Add Branch Office Tunnel panel appears.

4 Set the Tunnel name to Example.

5 Set the Local Endpoint to **10.10.11.2**.

6 Set the Remote Endpoint to **10.10.11.1**.

7 Set the Pre-shared key type to **Text**.

8 Set the Pre-shared key to **123**.

9 Click **OK** on the Add Branch Office Tunnel panel.

10 Select the Branch Office Tunnel you just created.
The details panel appears.

11 Click the **Local Accessible Networks** tab.

12 Click **Add**.
The Add Local Accessible Network panel appears.

13 Set the Local Accessible Networks to **14.14.14.0** with a subnet mask of **255.255.255.0**.

14 Click **OK.**

15 Click the **Remote Accessible Networks** tab.

16 Click **Add**.
The Add Remote Accessible Network panel appears.

17 Set the Remote Accessible Networks to **12.12.12.0** with a subnet mask of **255.255.255.0**.

18 Click **OK.**

IPSec Remote User Tunnel configuration

The IPSec Remote User feature allows remote users to dial in to an Internet Service Provider (ISP) anywhere in the world and connect to the corporate network in a secure way. All the remote user requires is an Contivity VPN client installed on their computer. This removes the need for the traditional corporate remote access environments where banks of modems were employed to handle incoming service requests.

BCM 4.0 has been tested with versions up to release V_06_01.014 of the Contivity VPN Client. To obtain a copy of this client software, contact your authorized Business Communications Manager 4.0 distributor. The VPN IPsec remote access user client software is a Windows application available for the latest releases of Windows 95, Windows 98, Windows NT Workstation, Windows NT Server, Windows 2000 and Windows XP. This client software comes with complete online Help.

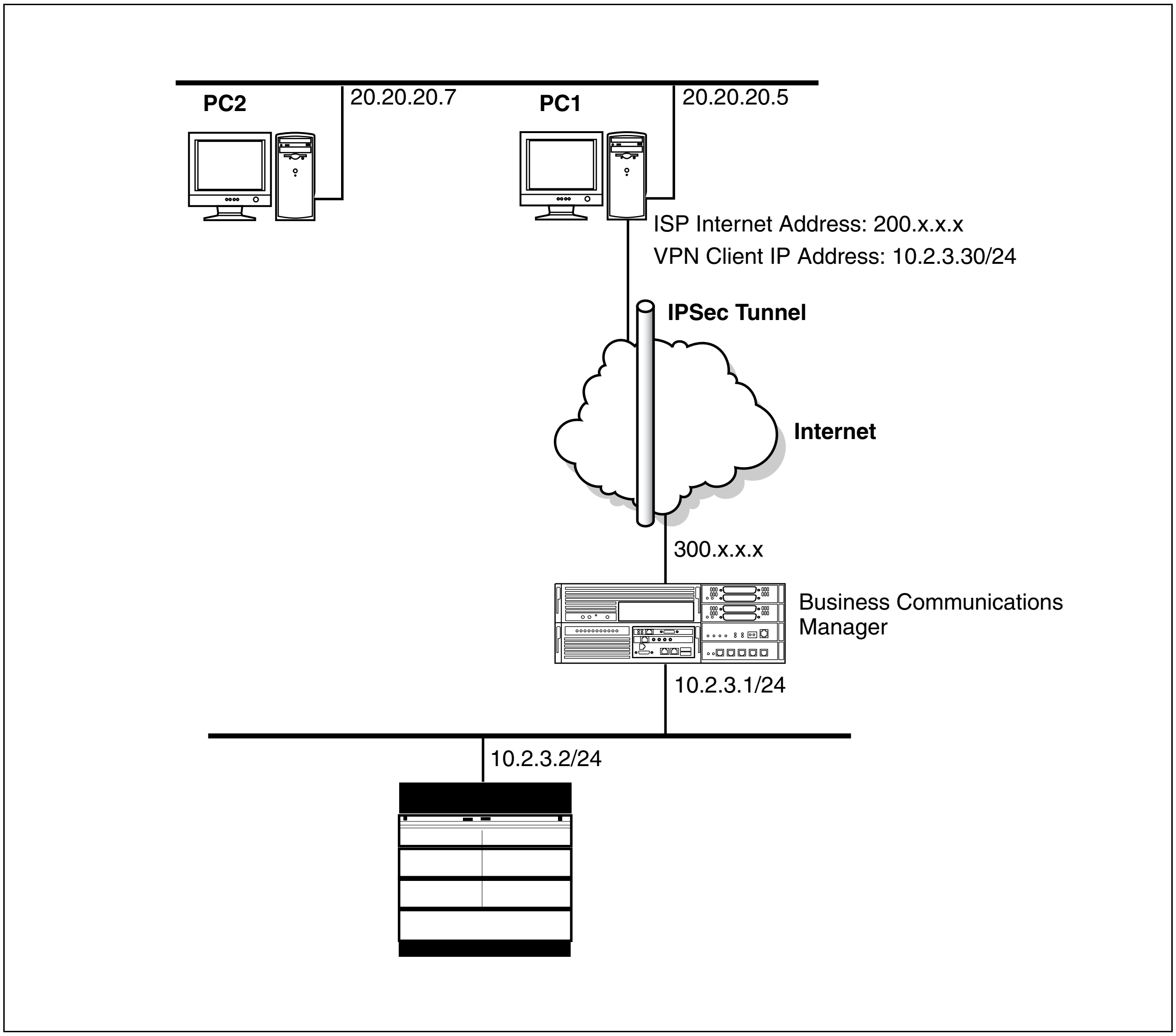
IPSec Remote User Tunnel Authentication

BCM only supports User Name and Password authentication from the VPN Client. The usernames/passwords are fully integrated with the BCM Groups, Accounts and Privileges section. The usernames/passwords are authenticated using the BCM local authentication functionality. No other form of Authentication is supported. BCM does not support Group ID authentication.

Split Tunneling

All client traffic is tunneled through the BCM by default. Split Tunneling allows you to configure specific network routes that are downloaded to the client. Only these network routes are then tunneled. Any other traffic goes to the local computer interface. Split tunneling allows you to print locally, for example, even while you are tunneled into the BCM.

Figure 206 Example of a Split Tunneling environment


PC2
20.20.20.7
PC1
20.20.20.5
ISP Internet Address: 200.x.x.x
VPN Client IP Address: 10.2.3.30/24
IPSec Tunnel
Internet
300.x.x.x
Business Communications Manager
10.2.3.1/24
10.2.3.2/24

In the example in the figure above, PC1 and PC2 are on a home IP network (20.20.20.0/255.255.255.0). PC1 is also connected to the Internet with an ISP granted IP address of 200.x.x.x. PC1 runs an IPSec VPN Client and connects to the BCM. BCM assigns this VPN Client connection an IP address of 10.2.3.30.

If Split tunneling is disabled, PC1 will NOT be able to access PC2 as ALL traffic will be sent down the IPSec tunnel.

However, if the Remote User Account has Split Tunneling enabled with split tunnel network IP addresses of 10.2.3.0/255.255.255.0, PC1 can establish an IPSec tunnel. When the client establishes an IPSec tunnel, this network address is loaded into the client application. PC1 can then access any system on the 10.2.3.0 network as well as accessing PC2 on IP network 20.20.20.0, while the VPN IPSec Client is still connected.

Split Tunneling security considerations

BCM takes precautions against violators potentially hacking tunneled information when the BCM is operating in Split Tunnel mode.

The primary precaution is to drop packets that do not have the IP address that is assigned to the tunnel connection as its source address. For example, if you have a PPP dial-up connection to the Internet with an IP address of 192.168.21.3, and you set up an IPSec client connection to a BCM and you are assigned an IPSec client IP address of 192.192.192.192, then any packets that attempt to pass through the IPSec client tunnel connection with a source IP address of 192.168.21.3 (or any address other than 192.192.192.192) will be dropped.

➡ **Note:** To completely eliminate security risks, you should not use the Split Tunneling feature.

Adding a Remote User IPSec Tunnel

A Remote User IPSec Tunnel connects a remote computer to the BCM system.

➡ **Note:** The remote computer must have the VPN Client installed.

➡ **Note:** If the computer running the VPN client is not on the same subnet as the Destination address (i.e. there is at least one router between the computer and the BCM), then the default Next Hop Router on the BCM must also be through this interface. For instructions on setting up a default Next Hop Router, refer to "Configuring Net Link Manager" on page 561.

Assigning an IP Address to a Remote User Account

The Remote User tunnel requires that an IP address is assigned to the Remote User when they log into the BCM. This IP address must be in the private IP network that the Remote User is able to access.

The BCM supports two methods of assigning an IP Address to the Remote User Tunnel. You can use a static IP address or a dynamic IP address from an IP Address Pool.

Static IP Address

To assign a static IP address to the Remote User account, you must configure the following two options when you configure the Remote User Tunnel settings:

- Static IP Address
- Subnet Mask

Dynamic IP address from an IP Address Pool

To assign a dynamic IP address, you must configure a Remote IP Address Pool and assign the Remote IP Address Pool to the Remote User Tunnel. For information about how to configure a Remote IP Address Pool, refer to "IP Address Pool" on page 676. To assign the Remote IP Address Pool to the Remote User Account, you must configure the IP Address Pool Name option when you configure the Remote User Account settings.

Note: You must configure either the **IP Address Pool Name** option or the **Static IP Address** and **Subnet Mask** options.

Note: When assigning IP addresses for Remote users, make sure that no conflicts can occur with IP Addresses already assigned on the private network. If the private network contains a DHCP server, the range assigned in the IP Address Pool or the Static IP Address must be excluded from the DHCP IP address range.

Adding a Remote User IPSec Tunnel involves the following:

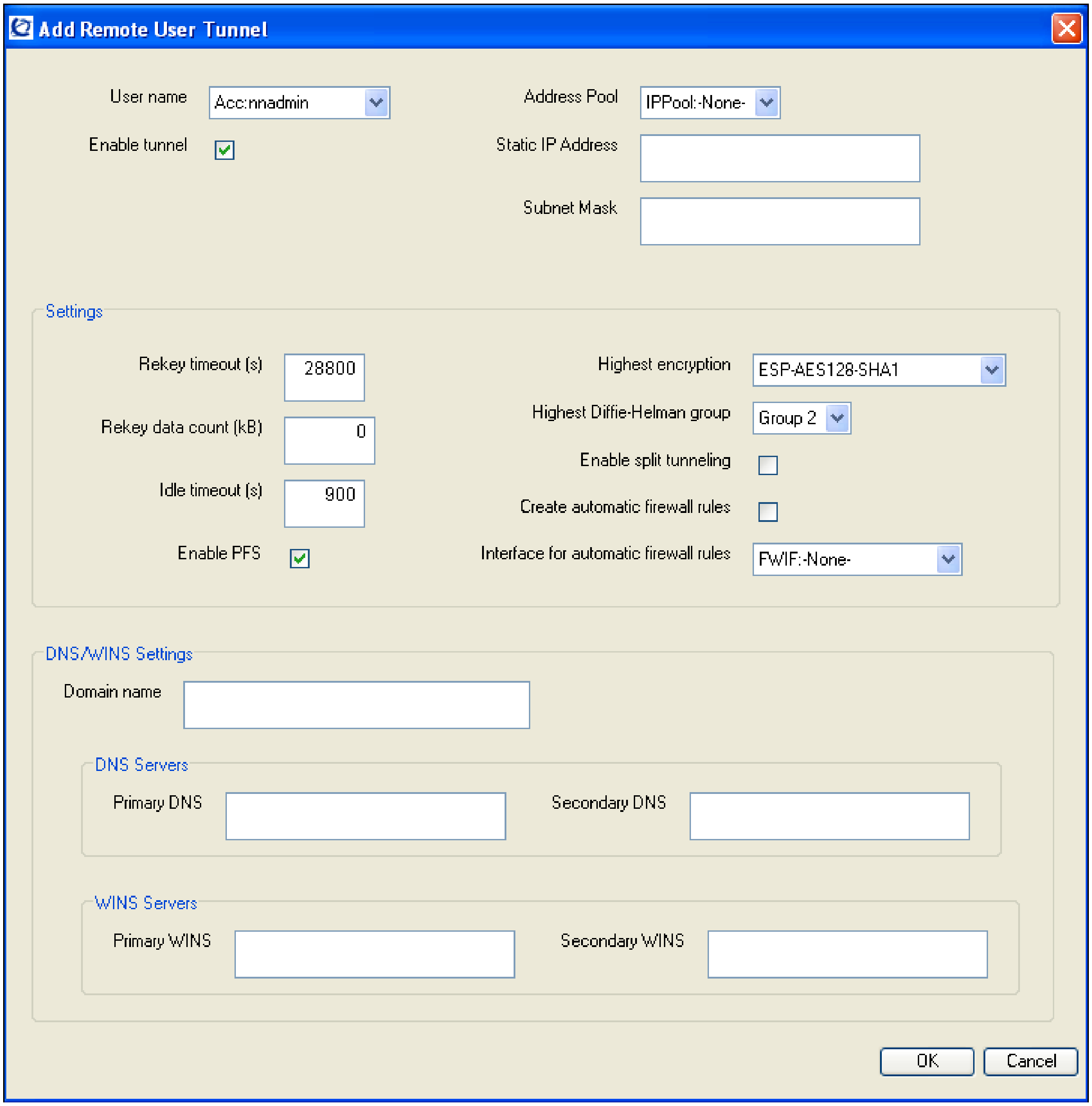
- "Remote User Tunnels" on page 670
- "IP Address Pool" on page 676

Remote User Tunnels

To add a remote user tunnel

1. Click **Configuration > Data Services > IPSec VPN**.
2. Click the **Remote User Tunnels** tab.
3. Click **Add**.
   The Add Remote User Tunnel panel appears. See Figure 207.
4. Configure the Remote User Tunnel settings. Refer to the information in Table 177.
5. Click **OK**.

Figure 207 Remote User Tunnel



User name: Acc:nnadmin
Enable tunnel: ☑
Address Pool: IPPool:-None-
Static IP Address:
Subnet Mask:

Settings

Rekey timeout (s): 28800
Rekey data count (kB): 0
Idle timeout (s): 900
Enable PFS: ☑
Highest encryption: ESP-AES128-SHA1
Highest Diffie-Helman group: Group 2
Enable split tunneling: □
Create automatic firewall rules: □
Interface for automatic firewall rules: FWIF:-None-

DNS/WINS Settings

Domain name:

DNS Servers

Primary DNS:
Secondary DNS:

WINS Servers

Primary WINS:
Secondary WINS:

Table 177 IPSec Remote User Tunnel settings (Sheet 1 of 3)

Attribute	Value	Description
User name	<alphanumeric>	Select the Remote User identifier from a list. The User Name uniquely identifies a Remote User. The User Name identifier does not have any significance, other than uniquely identifying an entry. Default: Acc:nnadmin
Enable tunnel	<check box>	Select to activate the tunnel.

Table 177 IPSec Remote User Tunnel settings (Sheet 2 of 3)

Attribute	Value	Description
Address Pool	<IP address pool>	Select the Remote IP Address Pool List you want to use for this Remote User Account. This allows you to assign a Dynamic IP Address (from the IP Address Pool) to the Remote User when they connect. **Note1**: If you select a Remote IP Address Pool List you do not have to specify the Static IP Address or the Static subnet mask. **Note2**: You must add a Remote IP Address Pool List before you can select it from the drop list.
Static IP Address	<IP address>	Specify the IP address that is used by the remote computer, if the remote computer is using a static IP address. **Note**: You do not need to enter a Static IP address if the Account is using a dynamic IP Address Pool.
Subnet Mask	<IP address>	Specify the Subnet Mask that is used by the remote computer, if the remote computer is using a static IP address. **Note**: You do not need to enter a Subnet Mask if the remote computer is using dynamic IP addressing.
Settings		
Rekey timeout (s)	<0-359999>	Specify the amount of time you can use a key before the tunnel is re-negotiated. You should limit the lifetime of a single key used to encrypt data or else you will compromise the effectiveness of a single session key. Use the Rekey Timeout setting to control how often new session keys are exchanged between servers. You cannot set the Rekey Timeout setting to less than three minutes, except to disable the timeout by entering 0. Default: 28800 secs. **Note:** A setting of 0 disables the Rekey Timeout setting.
Rekey data count (kB)	<0-1000000>	Specify the amount of data you can transmit on the tunnel before the tunnel is re-negotiated. Default: 28800 Kbytes **Note**:A setting of 0 disables the Rekey Data Count. **Note**: If you set the Rekey Data Count too low, the tunnel is re-negotiated too often and will consume extra system resources.
Idle timeout (s)	<0-359999>	Specify the amount of time the tunnel can remain idle before the tunnel is closed. You cannot set the Idle Timeout setting to less than three minutes, except to disable the timeout by entering 0. Default: 900 secs. **Note**: A setting of 0 disables the Idle timeout setting.
Enable PFS	<check box>	Enable Perfect Forward Secrecy (PFS). With PFS, keys are not derived from previous keys. This ensures that one key being compromised cannot result in the compromise of subsequent keys.

Table 177 IPSec Remote User Tunnel settings (Sheet 3 of 3)

Attribute	Value	Description
Highest encryption	ESP-AES128-SHA1 ESP-3DES-SHA1 ESP-3DES-MD5 ESP-DES56-SHA1 ESP-DES56-MD5 AH Authentication only (SHA1) AH Authentication only (MD5)	Select the highest encryption level allowed on this IPSec tunnel. When the encryption level is negotiated for this tunnel, BCM will not use any encryption level higher than the encryption level specified in this field. For a description of the encryption levels, refer to "Encryption" on page 644. Default: ESP-AES128-SHA1
Highest Diffie-Hellman	Group 5 Group 2 Group 1	Diffie-Hellman is a public-key cryptographic protocol that allows two parties to establish a shared secret over an insecure communications channel. It is also used within IKE (Internet Key Exchange) to establish session keys. IPSec uses the Diffie-Hellman algorithm to provide the keying material for all other encryption keys. Higher (larger bit keying material) Diffie-Hellman groups provide more security but require more processor time. Default: Group 2
Enable split tunneling	<check box>	Select to allow the remote computer to use Split Tunneling.
Create automatic firewall rules	<check box>	Select to create automatic firewall rules. Default: Cleared - which means that no rules are generated. See Figure 201 for an example of a remote user tunnel firewall configuration.
Interface for automatic firewall rules	<drop-down menu>	Select for which interface to generate Firewall Filter rules. These rules are necessary to allow packets for this Remote User tunnel through the firewall. Default: None - which means that no rules are generated.
DNS/WINS Settings		
Domain Name		Specify the Domain Name of the Domain in which the remote computer resides.
DNS Servers		
Primary DNS	<IP address>	Specify the IP address of the Primary DNS server that the remote computer uses.
Secondary DNS		Specify the IP address of the Secondary DNS server the remote computer uses. The remote computer uses the Secondary DNS server if the Primary DNS server is not available or does not have an entry for the domain name specified.
WINS Servers		
Primary WINS	<IP address>	Specify the IP address of the Primary WINS server that the remote computer uses.
Secondary WINS	<IP address>	Specify the IP address of the Secondary WINS server that the remote computer uses.

To modify a remote user tunnel

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Remote User Tunnels** tab.

3 Click the Remote User Tunnel you want to modify.
The details panel appears.

4 Click **Modify**.
The Modify Remote User Tunnel Panel appears. See Figure 207.

5 Change the required Remote User Tunnel settings.
Configure the Remote User Tunnel settings. Refer to the information in Table 177.

6 Click **OK**.

To delete a remote user tunnel

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Remote User Tunnels** tab.

3 Click the Remote User Tunnel you want to delete.
The details panel appears.

4 Click **Delete**.
A message prompts you to confirm the deletion. See Figure 207.

5 Click **Yes**.

Split Tunnel Networks

To add a split tunnel network

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Remote User Tunnels** tab.

3 Click the Remote User Account you want to modify.
The details panel appears.

4 Click the **Split Tunnel Networks** tab.

5 Click **Add**.
The Split Tunnel Network panel appears. See Figure 208.

6 Configure the Split Tunnel Network settings. Refer to the information in Table 178.

7 Click **OK**.

Figure 208 Split Tunnel Network

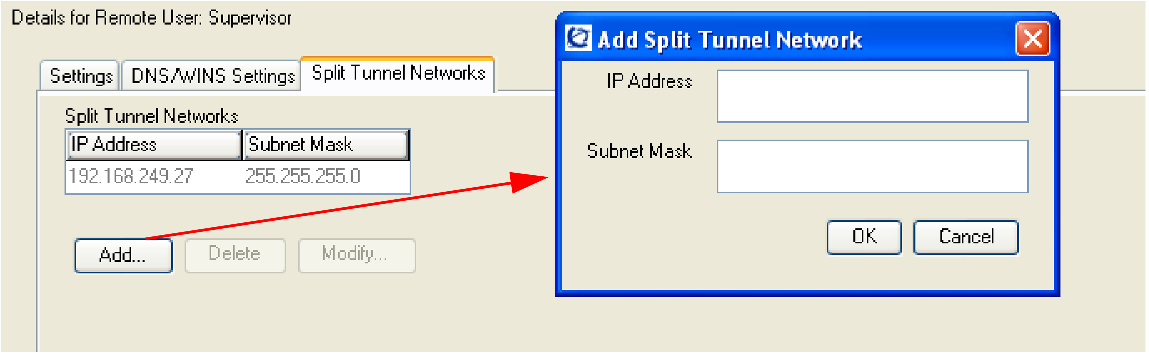


Table 178 Split Tunnel Network settings

Attribute	Value	Description
IP Address	<IP address>	Configure the specific IP network addresses that are routed through the IPSec Tunnel. All other IP traffic is routed in the normal fashion.
Subnet Mask	<IP address>	Specify the subnet mask for the other network.

To modify a split tunnel network

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Remote User Tunnels** tab.

3 Click the Remote User Account you want to modify.
The details panel appears.

4 Click the **Split Tunnel Networks** tab.

5 Click the Split Tunnel Network you want to modify.

6 Click the **Modify** button.
The Split Tunnel Network panel appears.

7 Change the required Split Tunnel Network settings.
For information about the settings. Refer to the information in Table 178.

8 Click **OK**.

To delete a split tunnel network

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **Remote User Tunnels** tab.

3 Click the Remote User Tunnel you want to modify.
The details panel appears.

4 Click the **Split Tunnel Networks** tab.
The Split Tunnel Networks panel appears.

5 Click the Split Tunnel Network you want to delete.

6 Click **Delete**.
A confirmation message appears.

7 Click **Yes**.

IP Address Pool

Remote access users who are using tunneling protocols require two IP addresses to form packets. The addresses are normally referred to as outer and inner addresses. The outer address, or public address, is visible when packets are traveling through the public data networks (PDNs). This address is negotiated between the client and the ISP to which it is connected. BCM does not have control of this address.

The inner IP address is the one that eventually appears on the private network when the outer layers of the packet are removed. Therefore, this address must lie within the private network address space. BCM provides the remote user with the inner IP address during tunnel setup. This address can come from a defined static IP address for this user account or from an internal address pool.

When assigning IP addresses for remote users, make sure that no conflicts can occur with IP addresses already assigned on the private network.

To add an IP address pool

1 Click **Configuration > Data Services > IPSec VPN**.

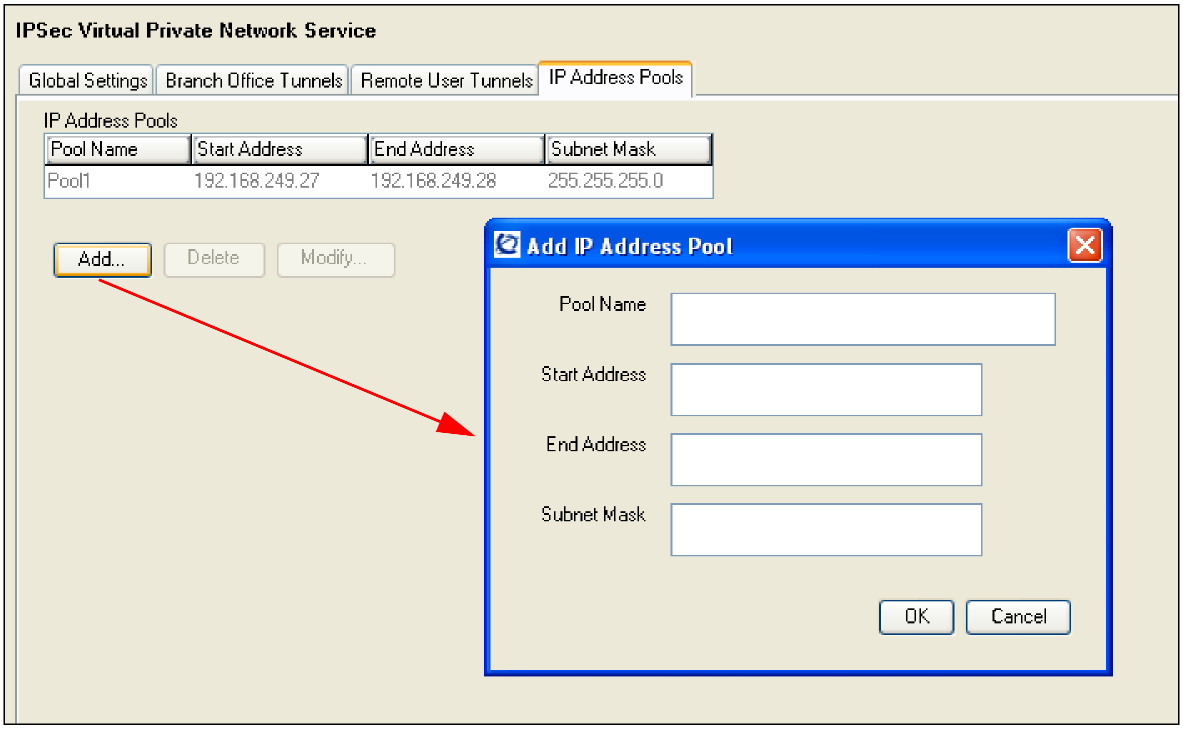
2 Click the **IP Address Pools** tab.

3 Click **Add.**
The IP Address Pool List panel appears. See Figure 209.

4 Configure the Remote IP Address Pool List settings. Refer to the information in Table 179.

5 Click **OK**.

Figure 209 IP address pool



IPSec Virtual Private Network Service

Pool Name	Start Address	End Address	Subnet Mask
Pool1	192.168.249.27	192.168.249.28	255.255.255.0

Table 179 IP Address Pool settings

Attribute	Value	Description
Pool Name		Specify the Remote IP Address Pool List identifier. The Pool Name uniquely identifies a Remote IP Address Pool List. If you specify an existing Pool Name, you receive an error message. The Remote IP Address Pool List identifier does not have any significance, other than uniquely identifying an entry.
Start Address	<IP address>	Specify the first IP address in the Remote IP Address Pool List.
End Address	<IP address>	Specify the last IP address in the Remote IP Address Pool List.
Subnet Mask	<IP address>	Specify the subnet mask for the Remote IP Address Pool List.

To modify an IP address pool

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **IP Address Pools** tab.
The IP Address Pool List panel appears.

3 Select the IP Address Pool List you want to modify.

4 Click **Modify**.
The IP Address Pool List panel appears. See Figure 209.

5 Modify the Remote IP Address Pool List parameters. Refer to the information in Table 179.

6 Click **OK**.

To delete an IP address pool

1 Click **Configuration > Data Services > IPSec VPN**.

2 Click the **IP Address Pools** tab.
The IP Address Pool List panel appears.

3 Click the IP Address Pool List you want to delete.

4 Click D**elete**.
A message prompts you to confirm the deletion. See Figure 209.

5 Click **Yes**.

Chapter 69
Configuring DNS

BCM can function as a DNS proxy or a DNS client.

The following path indicates where to configure the DNS settings in Element Manager:

- Element Manager: **Configuration > Data Services > DNS**

Refer to the following:

- "DNS proxy"
- "DNS client"

DNS proxy

When BCM receives DNS requests from clients, it first checks its local cache for name entries and records. If found locally, BCM immediately responds to clients. Otherwise, BCM creates a new DNS request to the remote Primary or Secondary DNS servers on behalf of the client. If the remote DNS server responds with the requested records, they are forwarded to clients and cached in BCM. By caching the DNS requests, the DNS proxy service on BCM reduces the number of external DNS requests and thus reduces the amount of WAN traffic.

The DNS proxy service also provides additional security. When DNS requests are sent to the Primary or Secondary DNS servers, the BCM published IP address is used for the request. By using the BCM IP address, the IP addresses of the internal users can remain hidden.

DNS client

If the DNS proxy is not enabled, then the BCM is configured as a DNS client through the Element Manager. This can be done in two locations:

- **Configuration > Data Services > DNS**
- **Configuration > Data Services > DHCP > Subnets > General Settings** tab

If this option is enabled, the DNS IP address for DNS clients will be obtained from the DHCP server. This will direct the DNS queries to another box, which has a DNS server.The DNS client settings from DHCP will be overritten by DNS if DNS Proxy is enabled or **Obtain settings from DHCP** is disabled.

Using the BCM DNS service

Consider the following guidelines when using DNS:

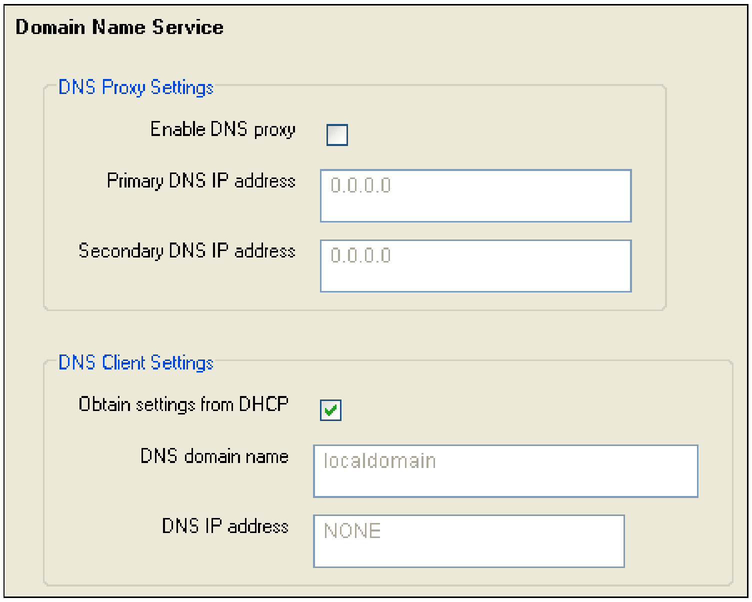
- If you enable the BCM DNS service, ensure that you configure each workstation on the network to use BCM as DNS server, and that the primary and secondary DNS server IP addressees are configured on the BCM. If the DNS server IP addresses are not set on the BCM, then DNS proxy is not functional.

- When you disable BCM DNS service, set the DNS Server field in DHCP configuration to the remote DNS server IP address. If DHCP service is also disabled in BCM, you must configure each workstation on your network to use the remote DNS server.
  - DHCP server enabled - DNS proxy enabled
  - DHCP server enabled - DNS proxy disabled
  - DHCP server disabled - DNS proxy enabled
  - DHCP server disabled - DNS proxy disabled

To configure DNS services settings

1. Click **Configuration > Data Services > DNS**.
   The Domain Name Server panel appears.
2. Configure the DNS Summary attributes. Refer to the information in Table 180.

Figure 210 Domain Name Service Settings



Domain Name Service

DNS Proxy Settings

Enable DNS proxy □

Primary DNS IP address 0.0.0.0

Secondary DNS IP address 0.0.0.0

DNS Client Settings

Obtain settings from DHCP ☑

DNS domain name localdomain

DNS IP address NONE

Table 180 Domain Name Server (Sheet 1 of 2)

Attribute	Value	Description
DNS Proxy Settings		
Enable DNS proxy	<check box>	Select to enable the BCM DNS cache proxy. Enabling the DNS proxy will override DNS IP address settings for DNS client from DHCP. Default: Disabled
Primary (and Secondary) DNS IP address	<IP address>	Specify the IP addresses of the primary DNS server and the secondary DNS server in a valid dot format. Range: 0.0.0.0-255.255.255.255 Default value: 0.0.0.0 **Note**: You must specify the IP addresses for the DNS proxy to function.
DNS Client Settings		
Obtain settings from DHCP	<check box>	Retrieve settings from the DHCP component. If enabled, the DNS IP address is obtained from the DHCP server, provided DHCP is enabled for the interface. If disabled, the DNS domain name and IP address can be configured for DNS client settings. **Note**: This field will be read-only if DNS Proxy is enabled or if DHCP is not enabled. DNS domain name and DNS IP address will be read-only if this option is enabled. Default: Enabled

Table 180 Domain Name Server (Sheet 2 of 2)

Attribute	Value	Description
DNS domain name		Specify the domain name that BCM and its DHCP clients uses. The domain is modified, the setting automatically copies to **Domain Name** global options under **DHCP** service. Maximum of 40 characters. (No spaces allowed). **Note**: This field will be read-only if DNS Proxy is enabled or if **Obtain Settings from DHCP** is enabled. Default: localdomain
DNS IP address		Specify the server name for DNS client functionality. **Note**: This field will be read-only if DNS Proxy is enabled or if **Obtain Settings from DHCP** is enabled. Default value is “NONE” if the server name is not present for DNS client.

Note: The DNS proxy carries security features because it keeps all of the published IP addresses from external web servers. For information on other security features, see “Configuring NAT (Network Address Translation)” on page 607 and “Configuring IP Filter Rules” on page 615.

Chapter 70
Configuring Web Cache

When you use BCM as a web proxy, BCM can store, or cache, information downloaded from the Internet. A proxy is a server that acts on behalf of another. Web caching allows LAN workstations to share common information downloaded from the Internet.

The Web Cache on the BCM supports the LAN, WAN, and DIALUP interfaces.

With BCM configured as a web proxy with web caching:

- LAN workstations have shorter download times.
- The system stores previously downloaded information for future use by all workstations on the LAN.
- BCM retrieves information from the Internet only if it is not already cached or if the cached file is out of date compared to the information on the Internet.

To use the Web Cache on the BCM, you must configure the client computer to use an Internet Proxy at port 6800, where the internet proxy is the BCM.

The web proxy also provides security features similar to the DNS proxy. It hides all of the internal browsers' IP addresses from external web servers. External web servers see only the BCM IP address.

Guidelines for using Web caching/Proxy

The BCM web proxy uses a web server for running in HTTP-Proxy mode.

Security Note: When Web Cache is enabled, the proxy service is also enabled on all of the network interfaces. If your BCM is connected to an external interface, it is possible for someone on the external network to use the proxy service to access one of the internal networks.

To block the proxy service on any of the network interfaces, select the check box beside the interface you want to block in the **Proxy Blocked Interfaces** field.

The default for the **Proxy Blocked Interfaces** field is selected (blocked) for the LAN1 interface.

When Web Cache is disabled, the proxy service is also disabled on all the network interfaces

Refer to "Important Web Cache considerations" on page 686 for more details.

Consider the following guidelines when using web caching/proxy:

- You cannot use the web server installed on BCM as a general purpose HTTP server. It is only used by the BCM web-based management client and Web Cache services.

- If you want to run web sites on your network, you must have a separate HTTP server running on a system other than the BCM system. There are two options available for the IP address you publish for your website. You can publish a separate IP address for the HTTP server or you can publish the same IP address as your BCM.
  To publish a separate IP address for the HTTP server, publish the IP address of the computer on which you are running the HTTP server.
  To publish the same IP address used for BCM, set up a NAT rule to change the public address of the HTTP server to the IP address of BCM.
- Some secure web sites are not accessible through the BCM Web Cache service. If you are having problems accessing a secure web site, turn off the Web Cache service and try again.
- Web Cache is disabled by default.

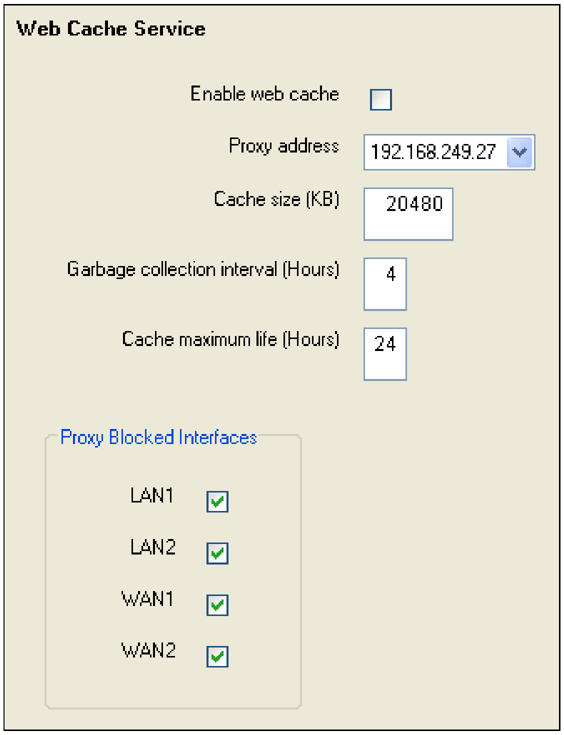
To configure Web Cache settings

1 Click **Configuration > Data Services > Web Cache**.
The **Web Cache Service** panel appears.

2 Configure the Web Cache attributes according to Table 27.

Note: Refer to “Important Web Cache considerations” on page 686 for more information on the Web Cache attributes.

Figure 211 Web Cache Service



Web Cache Service

Enable web cache □

Proxy address 192.168.249.27

Cache size (KB) 20480

Garbage collection interval (Hours) 4

Cache maximum life (Hours) 24

Proxy Blocked Interfaces

LAN1 ☑

LAN2 ☑

WAN1 ☑

WAN2 ☑

Table 27 Web Cache attributes

Attribute	**Value**	**Description**
Enable web cache	<check box>	Enable or disable the web proxy in BCM. Default: Disabled
Proxy address	<IP address>	Specify which IP address to use for interacting with HTTP clients. Since BCM typically has more than one IP interface and associated IP Address, users can choose this value. Default: 0.0.0.0
Cache size (KB)	<1-100,000>	Specify the maximum size of the cache. Default: 20,480 KB
Garbage collection interval (Hours)	<1-24>	Specify the interval, in hours, between garbage collection operations on the cache. Default: 4 hours
Cache maximum life (Hours)	<1-24 >	Specify the maximum life, in hours, on the proxy server for cached HTTP pages. Default: 24 hours
Proxy Blocked Interfaces	LAN1 LAN2 WAN1 WAN2	Select the check boxes beside the interfaces on which you want BCM to block the Proxy service. Default: All interfaces are selected.

Important Web Cache considerations

Ensure that you consider the following items when configuring Web Cache:

Web Cache page update required

If you change the IP address for the LAN, WAN, or DIALUP interfaces, you must update the IP address on the Web Cache page and reset the Proxy Blocked Interfaces. If you do not reset the Web Cache attributes and Proxy Blocked Interfaces, the Web Cache will not function correctly.

DIALUP interfaces blocking condition

Proxy Blocking of DIALUP interfaces works only if the two BCMs are connected back-to-back using the DIALUP interface.

Additional IP addresses are not blocked

If you designate additional IP addresses for a supported interface, the additional IP addresses are not blocked when Proxy Blocked Interfaces is set to that interface. In this case, only the main IP address is blocked.

Chapter 71
Configuring QoS (Quality of Service) Queuing

Outbound traffic queuing is configured on a per-interface basis. The BCM queuing strategy follows the Nortel recommended model for seven traffic queue classes.

The premium service class (queue 1) uses a Strict Priority scheduler. This class is shaped so that other classes do not get starved. The shaping of the premium service class is based on a percentage (from 0% to 90%) of the available bandwidth of the interface. The unused bandwidth of the interface is distributed across the other service classes. Therefore, the other service classes get a minimum of 10?% of the total bandwidth of the interface and a maximum of up to 100%.

Service classes (queue 2 to 7) other than premium use a simpler variant of Weighted Fair Queuing (WFQ) called Hierarchical Token Buckets (HTB). The scheduler allows each service class to use a guaranteed minimum traffic rate, and allows unused bandwidth of other classes in the HTB scheduler to be redistributed to other classes.

The guaranteed bandwidth percentage value for service classes (queue 2 to 7) other than premium is based on the remaining unused bandwidth portion of the premium service class. So the sum of percentage values for the queue 2 to 7 must be equal to 100%.

The excess bandwidth percentage value applies to service classes other than premium. It must be set to a value greater or equal to guaranteed bandwidth, and can be set up to 100% per class. This value is used for traffic shaping on a per class basis.

Queuing

Refer to the following procedures:

- Configuring QoS Queuing
- To modify Queue Settings

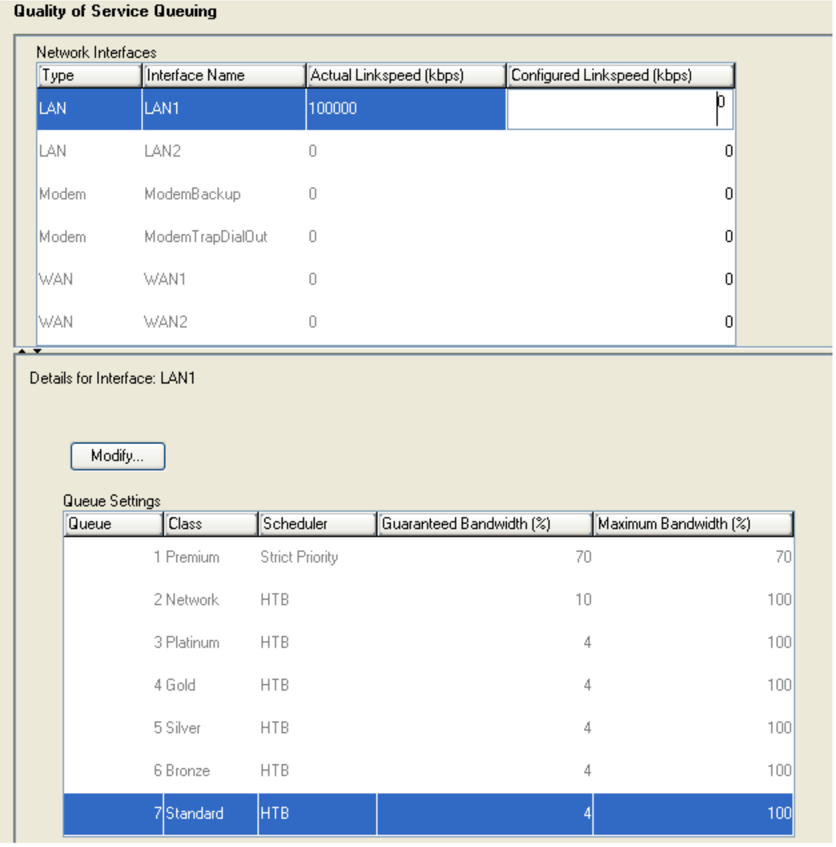
Configuring QoS Queuing

To configure QoS Queuing

1. Click **Configuration > Data Services > QoS Queuing.**
2. Click the interface you want to configure. For example: **LAN1**.
   The details panel appears. See Figure 212.
3. Configure the settings according to Table 28.

Note: Double click the Configured Link Speed (kbps) box to edit.

Figure 212 QoS Queuing LAN Details



Quality of Service Queuing

Network Interfaces

Type	Interface Name	Actual Linkspeed (kbps)	Configured Linkspeed (kbps)
LAN	LAN1	100000	0
LAN	LAN2	0	0
Modem	ModemBackup	0	0
Modem	ModemTrapDialOut	0	0
WAN	WAN1	0	0
WAN	WAN2	0	0

Details for Interface: LAN1

Queue Settings

Queue	Class	Scheduler	Guaranteed Bandwidth (%)	Maximum Bandwidth (%)
1	Premium	Strict Priority	70	70
2	Network	HTB	10	100
3	Platinum	HTB	4	100
4	Gold	HTB	4	100
5	Silver	HTB	4	100
6	Bronze	HTB	4	100
7	Standard	HTB	4	100

Table 28 QoS Queuing Settings (Sheet 1 of 2)

Attribute	Value	Description
Type	Read-only LAN Modem WAN	This represents the type of interface. This field is read-only.
Interface Name	Read-only	This represents the interfaces available for queuing in the BCM.

Table 28 QoS Queuing Settings (Sheet 2 of 2)

Attribute	Value	Description
Actual Link Speed (kbps)	Read-only	This attribute represents the actual outbound link speed in kbps that the BCM uses for the interface. When the interface comes up, this value is detected automatically by the BCM. If this attribute contains zero, then the interface is not yet up. If the BCM could not detect the outbound link speed automatically, then the actual link speed is displayed as “unknown” and the default outbound link speed is 1000 kbps (1 Mbps). In this case, the user must enter a value in the Configured link speed field in order to properly deal with traffic shaping and traffic distribution over the seven queues while maximizing bandwidth utilization. A known case of this is a WAN interface running in DTE mode.
Configured Link Speed (kbps)		This attribute represents the outbound link speed in kbps that the user configured for the interface. This value is only used by the BCM if the outbound link speed could not be detected automatically by the BCM. Default: 0.

To modify Queue Settings

1 Click **Configuration > Data Services > QoS Queuing.**

2 Click the interface you want to configure.
The details panel appears.

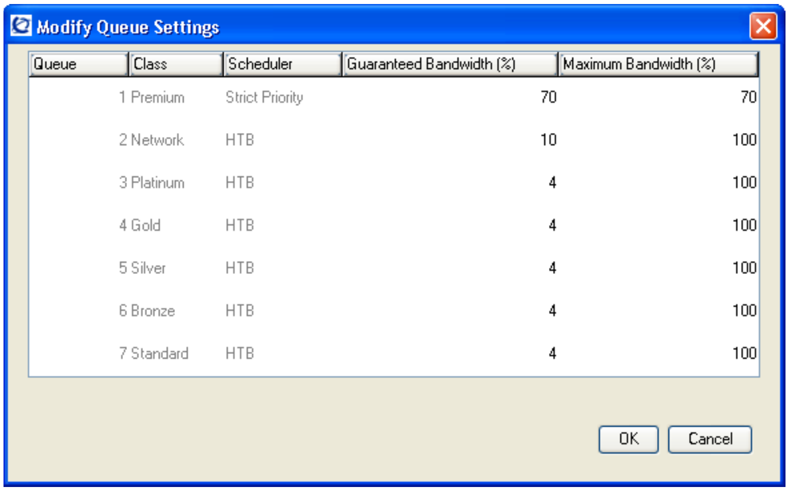
3 Click the **Modify** button
The Modify Queue Settings panel appears. See Figure 213.

4 Configure the Queue settings. Refer to the information in Table 213.

➡ **Note:** Double-click the box you want to modify to edit.

5 Click **OK**.

Figure 213 Modifying Queue Settings



Queue	Class	Scheduler	Guaranteed Bandwidth (%)	Maximum Bandwidth (%)
1	Premium	Strict Priority	70	70
2	Network	HTB	10	100
3	Platinum	HTB	4	100
4	Gold	HTB	4	100
5	Silver	HTB	4	100
6	Bronze	HTB	4	100
7	Standard	HTB	4	100

Table 29 Modifying Queue Settings

Attribute	Value	Description
Queue	Read-only 1-7	This is a fixed queue number on a per-service class.
Class	Read-Only Premium Network Platinum Gold Silver Bronze Standard	This is the fixed service class type of each queue.
Scheduler	Read-Only Strict Priority HTP	This represents the type each scheduler used by each service class.

Table 29 Modifying Queue Settings

Attribute	Value	Description
Guaranteed Bandwidth (%)	<numeric>	For the premium service class (queue 1), Guaranteed Bandwidth represents the percentage of the total available outbound link speed that can be used for forwarding premium traffic. Any traffic in excess of this value will be shaped. For service classes (queue 2 to 7) other than premium, Guaranteed Bandwidth represents the percentage of the traffic distribution in the Hierarchical Token Buckets (HTB) scheduler classes. The Guaranteed Bandwidth percentage value is based on the remaining unused bandwidth portion of the premium service class. The sum of percentage values for queue 2 to 7 must be equal to 100%.
Maximum Bandwidth (%)	<numeric>	Maximum bandwidth represents the amount of unused bandwidth that can be borrowed from other queues. It is represented by a percentage value of the total available outbound link speed that the interface can use to forward traffic. For the premium service class (queue 1), the default value is set to the Guaranteed bandwidth value. This ensures that traffic shaping takes place and prevents starvation of other queues. The unused bandwidth (shaped or not) is given to the remaining queues of this interface. For service classes (queue 2 to 7) other than premium, the default value is set to 100%. This ensures that unused bandwidth from other queues is used if needed. The default values on a per-queue basis are follows: Queue 1: 70% Queue 2: 100% Queue 3: 100% Queue 4: 100% Queue 5: 100% Queue 6: 100% Queue 7: 100%

Chapter 72
VLAN overview

A virtual LAN (VLAN) is a logical grouping of ports, controlled by a switch, and end-stations, such as IP telephones, configured so that all ports and end-stations in the VLAN appear to be on the same physical (or extended) LAN segment even though they may be geographically separated. VLAN IDs are determined by how the VLAN switch is configured. If you are not the network administrator, you must ask whoever manages the switch what the VLAN ID range is for your system.

VLANs aim to offer the following benefits:

- VLANs are supported over all IEEE 802 LAN MAC protocols, and over shared media LANs as well as point-to-point LANs.
- VLANs facilitate easy administration of logical groups of stations that can communicate as if they were on the same LAN. They also facilitate easier administration of move, add, and change in members of these groups.
- Traffic between VLANs is restricted. Bridges forward unicast, multicast, and broadcast traffic only on LAN segments that serve the VLAN to which the traffic belongs.
- For IP telephony, VLANs provide a useful technique to separate and prioritize the telephony traffic for L2 switches.
- VLAN also provides a shield from malicious traffic that may be targeted at the IP phone in order to steal or disrupt service.
- Reuse IP addresses in different VLANs.
- As far as possible, VLANs maintain compatibility with existing bridges and end stations.
- If all bridge ports are configured to transmit and receive untagged frames, bridges will work in plug-and-play ISO/IEC 15802-3 mode. End stations are able to communicate throughout the Bridged LAN.

Choosing DHCP for VLAN

By using the BCM DHCP server, you can configure DHCP to auto-assign a VLAN ID to each IP telephone that registers. With this configuration, you can also choose to manually enter VLAN IDs, if you choose. The BCM DHCP server becomes the default VLAN that everyone can reach. The server provides the network configuration information in the default VLAN, and it also provides the VLAN information for the network.

Specifying the site-specific options for VLAN

The BCM DHCP server resides in default VLAN and is configured to supply the VLAN information to the IP phones. The DHCP server will supply site-specific option in the DHCP offer message.

The following definition describes the Nortel i2004 specific, Site Specific option. This option uses the **reserved for site specific use** DHCP options (DHCP option values 128 to 254) and must be returned by the DHCP server as part of each DHCP OFFER and ACK message for the i2004 to accept these messages as valid. The i2004 will pull the relevant information out of this option and use it to configure the IP phone.

Format of field is: Type, Length, Data.

Type (1 octet):

- Five choices 0x80, 0x90, 0x9d, 0xbf, 0xfb (128, 144, 157, 191, 251).
- Providing a choice of five types allows the i2004 to work in environments where the initial choice may already be in use by a different vendor. Pick only one TYPE byte.

Length (1 octet): (variable depends on the message content)

Data (length octets):

- ASCII based
- format: `VLAN-A:XXX,YYY.ZZZ.`

  where `VLAN-A:` uniquely identifies this as the Nortel DHCP VLAN discovery.

  — `-A` signifies this version of this spec. Future enhancements could use `-B`, for example.

  — ASCII `,` (comma) is used to separate fields.

  — ASCII `.` (period) is used to signal end of structure.

  — `XXX`, `YYY` and `ZZZ` are ASCII-encoded decimal numbers with a range of 0-4095. The number is used to identify the VLAN Ids. A maximum of 10 VLAN Ids can be configured. `NONE` means no VLAN (default VLAN).

The DHCP Offer message carrying VLAN information has no VLAN tag when it is sent out from the DHCP server. However, a VLAN tag is added to the packet at the switch port. The packets are untagged at the port of the IP phone.

Appendix A

Silence suppression

This following describes using silence suppression on half-duplex and full-duplex links:

Silence suppression, also known as voice activity detection, reduces bandwidth requirements by as much as 50 per cent. The following explains how silence suppression functions on a Business Communications Manager 4.0 network.

G.711 and G.729, support Silence suppression.

A key to VoIP Gateways in business applications is reducing WAN bandwidth use. Beyond speech compression, the best bandwidth-reducing technology is silence suppression, also known as Voice Activity Detection (VAD). Silence suppression technology identifies the periods of silence in a conversation, and stops sending IP speech packets during those periods. Telco studies show that in a typical telephone conversation, only about 36% to 40% of a full-duplex conversation is active. When one person talks, the other listens. This is half-duplex. There are important periods of silence during speaker pauses between words and phrases. By applying silence suppression, average bandwidth use is reduced by the same amount. This reduction in average bandwidth requirements develops over a 20-to-30-second period as the conversation switches from one direction to another.

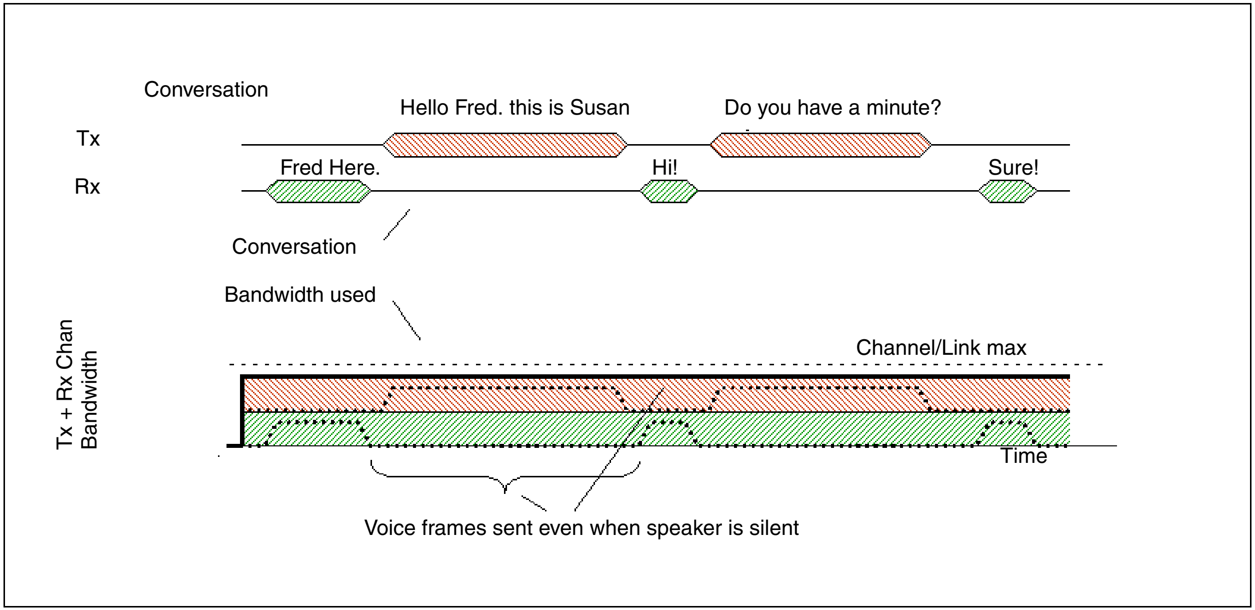
When a voice is being transmitted, it uses the full rate or continuous transmission rate.

The effects of silence suppression on peak bandwidth requirements differ, depending on whether the link is half-duplex or full-duplex.

Silence suppression on half-duplex links

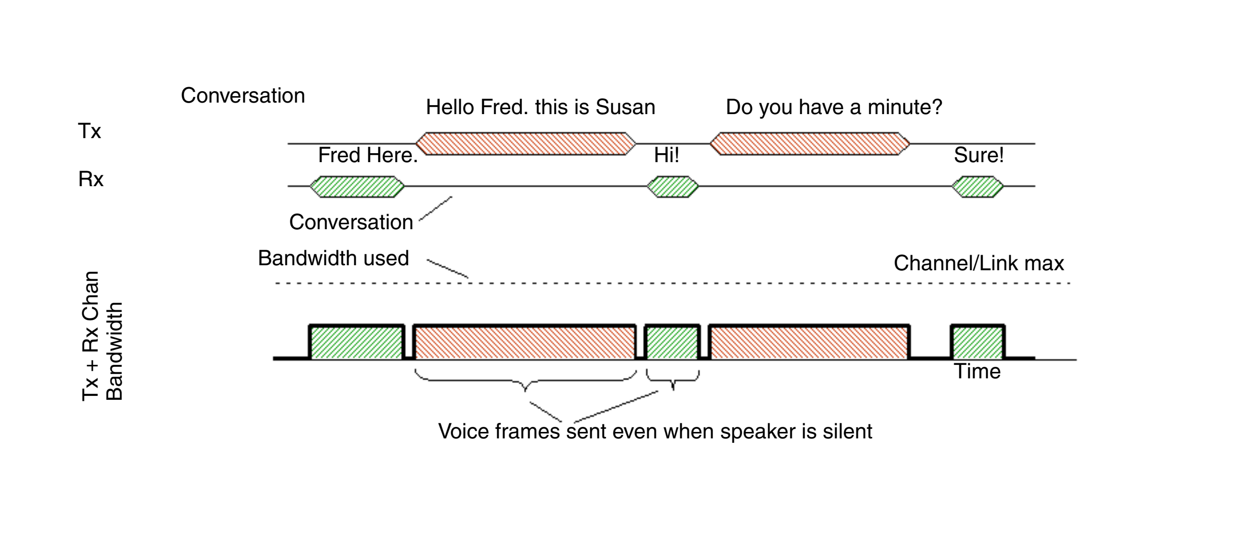
The following figure shows the bandwidth requirement for one call on a half-duplex link without silence suppression. Since the sender and receiver share the same channel, the peak bandwidth is double the full transmission rate. Because voice packets are transmitted even when a speaker is silent, the average bandwidth used is equal to the full transmission rate.

Figure 214 One call on a half-duplex link without silence suppression



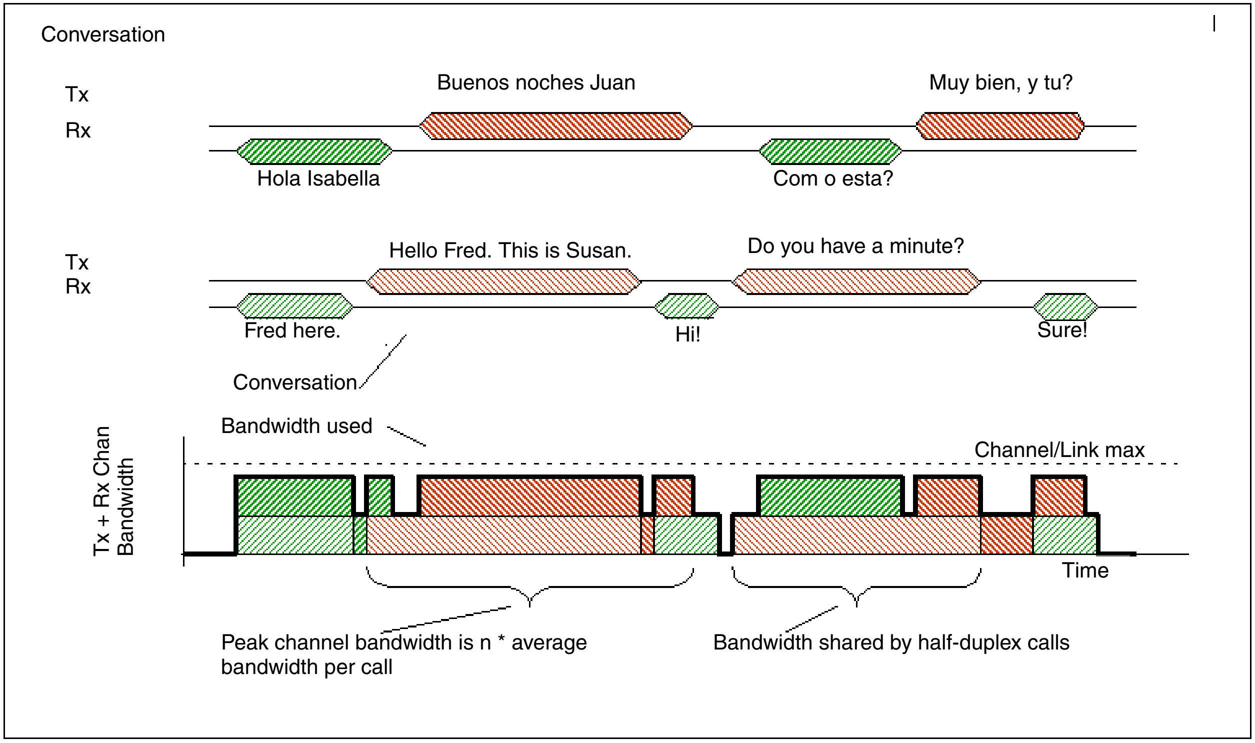
When silence suppression is enabled, voice packets are only sent when a speaker is talking. In a typical voice conversation, while one speaker is talking, the other speaker is listening – a half-duplex conversation. The following figure shows the peak bandwidth requirements for one call on a half-duplex link with silence suppression enabled. Because the sender and receiver alternate the use of the shared channel, the peak bandwidth requirement is equal to the full transmission rate. Only one media path is present on the channel at one time.

Figure 215 One call on a half-duplex link with silence suppression



The effect of silence suppression on half-duplex links is, therefore, to reduce the peak and average bandwidth requirements by approximately 50% of the full transmission rate. Because the sender and receiver are sharing the same bandwidth, this effect can be aggregated for a number of calls. The following figure shows the peak bandwidth requirements for two calls on a half-duplex link with silence suppression enabled. The peak bandwidth for all calls is equal to the sum of the peak bandwidth for each individual call. In this case, that is twice the full transmission rate for the two calls.

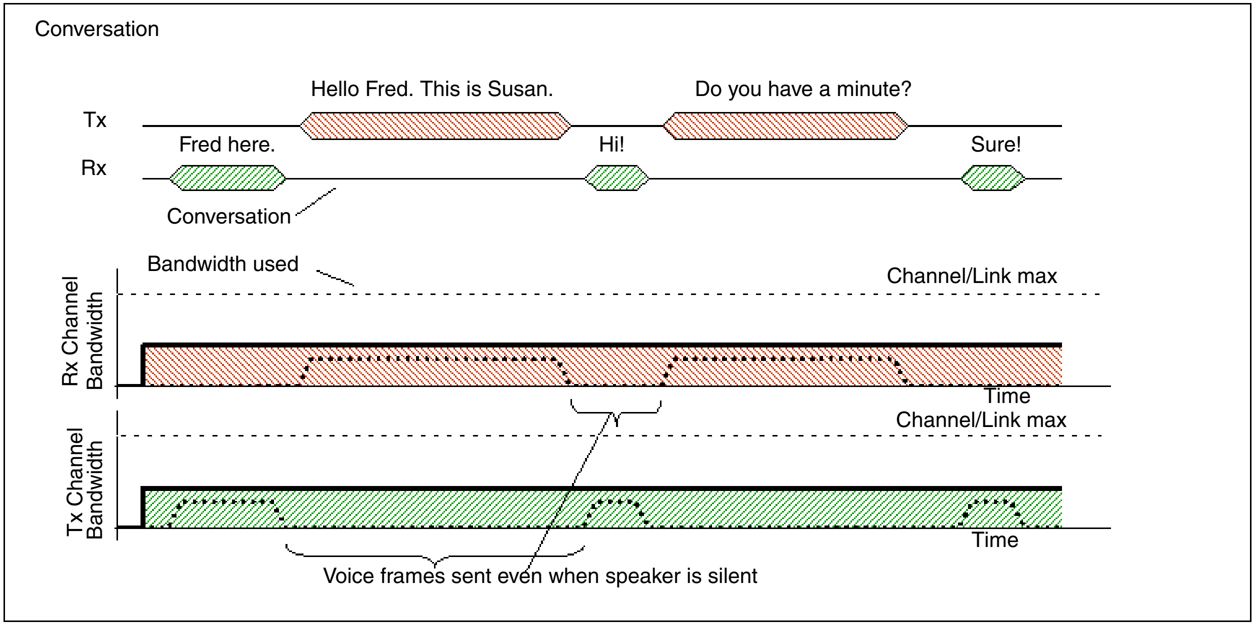
Figure 216 Two calls on a half-duplex link with silence suppression



Silence suppression on full-duplex links

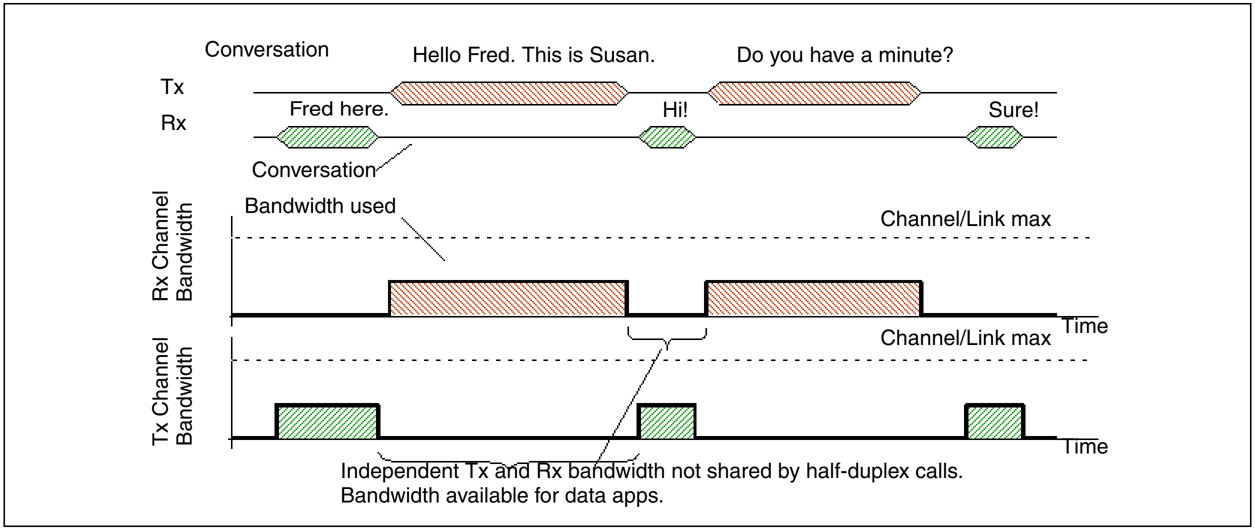
On full-duplex links, the transmit path and the receive path are separate channels, with bandwidths usually quoted in terms of individual channels. The following figure shows the peak bandwidth requirements for one call on a full-duplex link without silence suppression. Voice packets are transmitted, even when a speaker is silent. Therefore, the peak bandwidth and the average bandwidth used equals the full transmission rate for both the transmit and the receive channel.

Figure 217 One call on a full-duplex link without silence suppression



When silence suppression is enabled, voice packets are only sent when a speaker is talking. When a voice is being transmitted, it uses the full-rate transmission rate. Since the sender and receiver do not share the same channel, the peak bandwidth requirement per channel is still equal to the full transmission rate. The following figure shows the peak bandwidth requirements for one call on a full-duplex link with silence suppression enabled. The spare bandwidth made available by silence suppression is used for lower-priority data applications that can tolerate increased delay and jitter.

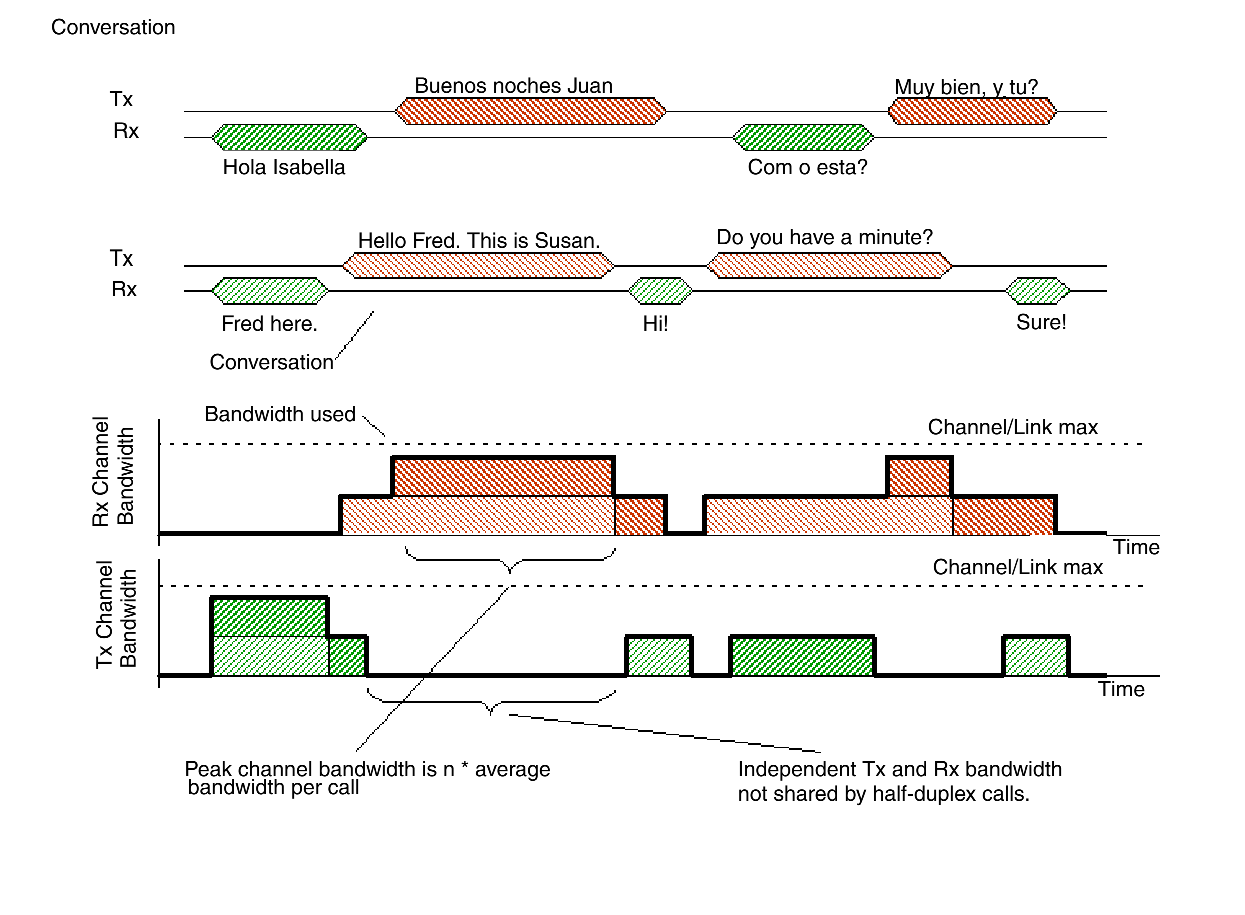
Figure 218 One call on a full-duplex link with silence suppression



When several calls are made over a full-duplex link, all calls share the same transmit path and they share the same receive path. Since the calls are independent, the peak bandwidth must account for the possibility that all speakers at one end of the link may talk at the same time. Therefore, the peak bandwidth for n calls is n * the full transmission rate. The following figure shows the peak bandwidth requirements for two calls on a full-duplex link with silence suppression. Note that the peak bandwidth is twice the full transmission rate, even though the average bandwidth is considerably less.

The spare bandwidth made available by silence suppression is available for lower priority data applications that can tolerate increased delay and jitter.

Figure 219 Two calls on a full-duplex link with silence suppression



Comfort noise

To provide a more natural sound during periods of silence, comfort noise is added at the destination gateway when silence suppression is active. The source gateway sends information packets to the destination gateway informing it that silence suppression is active and describing what background comfort noise to insert. The source gateway only sends the information packets when it detects a significant change in background noise.

Chapter 73
ISDN overview

The following provides some general information about using ISDN lines on your BCM system. Detailed information about ISDN is widely available through the internet. Your service provider can also provide you with specific information to help you understand what suits your requirements.

Refer to the following topics for information:

- “Welcome to ISDN”
- “Services and features for ISDN BRI and PRI” on page 703
- “ISDN hardware” on page 707
- “ISDN standards compatibility” on page 710
- “Planning your ISDN network” on page 711
- “Supported ISDN protocols” on page 712

Welcome to ISDN

Integrated Services Digital Network (ISDN) technology provides a fast, accurate and reliable means of sending and receiving voice, data, images, text, and other information through the telecom network.

ISDN uses existing analog telephone wires and multiplex it into separate digital channels which increases bandwidth.

ISDN uses a single transport to carry multiple information types. What once required separate networks for voice, data, images, or video conferencing is now combined onto one common high-speed transport.

Refer to the following topics:

- “Types of ISDN service” on page 702
- “ISDN layers” on page 702
- “ISDN bearer capability” on page 703

Analog versus ISDN

ISDN offers significantly higher bandwidth and speed than analog transmission because of its end-to-end digital connectivity on all transmission circuits. Being digital allows ISDN lines to provide better quality signaling than analog POTS lines, and ISDN out-of band data channel signaling offers faster call set up and tear down.

While an analog line carries only a single transmission at a time, an ISDN line can carry one or more voice, data, fax, and video transmissions simultaneously.

An analog modem operating at 14.4K takes about 4.5 minutes to transfer a 1MB data file and a 28.8K modem takes about half that time. Using one channel of an ISDN line, the transfer time is reduced to only 1 minute and if two ISDN channels are used, transfer time is just 30 seconds.

When transmitting data, the connect time for an average ISDN call is about three seconds per call, compared to about 21 seconds for the average analog modem call.

Types of ISDN service

Two types of ISDN services (lines) are available: Basic Rate Interface (BRI) and Primary Rate Interface (PRI). Each line is made up of separate channels known as B and D channels which transmit information simultaneously.

- BRI is known as 2B+D because it consists of two B-channels and one D-channel.
- PRI is known as 23B+D(in North America) or as 30B+D (in Europe). In North America, 23B+D consists of 23 B-channels and one D-channel (T1 carrier). In Europe, 30B+D consists of 30 B-channels and one D-channel (E1 carrier).

B-channels: B-channels are the bearer channel and are used to carry voice or data information and have speeds of 64 kbps. Since each ISDN link (BRI or PRI) has more than one B-channel, a user can perform more than one transmission at the same time, using a single ISDN link.

D-channels: The standard signaling protocol is transmitted over a dedicated data channel called the D-channel. The D-channel carries call setup and feature activation information to the destination and has speeds of 16 kbps (BRI) and 64 kbps PRI. Data information consists of control and signal information and for BRI only, packet-switched data such as credit card verification.

ISDN layers

ISDN layers refer to the standards established to guide the manufacturers of ISDN equipment and are based on the OSI (Open Systems Interconnection) model. The layers include both physical connections, such as wiring, and logical connections, which are programmed in computer software.

When equipment is designed to the ISDN standard for one of the layers, it works with equipment for the layers above and below it. There are three layers at work in ISDN for BCM. To support ISDN service, all three layers must be working properly.

- Layer 1: A physical connection that supports fundamental signaling passed between the ISDN network (your service provider) and the BCM system. When the LED on a BRI S/T Media Bay Module configured as BRI is lit, your layer 1 is functioning.
- Layer 2: A logical connection between the central office or the far end and the BCM system. BCM has one or two of these connections for each BRI link, and one for each PRI link. Without Layer 2, call processing is not possible.

- Layer 3: Also a logical connection between the ISDN network (your service provider) and the BCM system. For BRI lines, layer 3 is where call processing and service profile identifier (SPID) information is exchanged. This controls which central office services are available to the connection. For example, a network connection can be programmed to carry data calls.

➡ **Note:** Throughout this chapter, references are made to Service profile identifiers (SPIDs). SPIDs are a part of the BRI National ISDN standard. SPIDs are not used in the ETSI BRI standard or on PRI.

The three layers mentioned above is important when you are installing, maintaining, and troubleshooting an ISDN system. For information about troubleshooting ISDN, see the *BCM 4.0 Administration Guide* (N0060598).

ISDN bearer capability

Bearer capability describes the transmission standard used by the BRI or PRI line so that it can work within a larger ISDN hardware and software network.

The bearer capability for BRI and PRI is voice/speech, 3.1 kHz audio (fax), and data (unrestricted 64 kbps, restricted 64 kbps, or 56 kbps).

Services and features for ISDN BRI and PRI

As part of an ISDN digital network, your system supports enhanced capabilities and features, including:

- faster call set up and tear down
- high quality voice transmission
- dial-up Internet and local area network (LAN) access
- video transmission
- network name display
- name and number blocking (PRI, BRI and analog)
- access to public protocols

Refer to the following for additional information on features and services:

- “Network name display” on page 705
- “Name and number blocking (ONN)” on page 705
- “Call by Call Service Selection for PRI” on page 705
- “Emergency 911 dialing” on page 706
- “2-way DID” on page 707
- “Dialing plan and PRI” on page 707

PRI services and features

The services and features provided over PRI lines include:

- Call-by-call service selection (NI protocol)
- Emergency 911 dialing, internal extension number transmission
- access to Meridian 1 private networking (SL-1 protocol)

BRI services and features

The services and features provided over BRI lines include:

- data transmission at speeds up to 128 kbps per loop (depending on the bandwidth supported by your service provider)
- shared digital lines for voice and data ISDN terminal equipment

BCM Basic Rate Interface (BRI) also support D-channel packet service between a network and terminal connection. This allows you to add applications such as point-of-sale terminals (POSTA) without additional network connections. Connecting a POSTA allows transaction terminals (devices where you swipe credit or debit cards) to transmit information using the D channel of the BRI line, while the B channels of the BRI line remain available for voice and data calls. A special adapter links transaction equipment, such as cash registers, credit card verification rigs, and point-of-sale terminals, to the X.25 network, which is a data communications network designed to transmit information in the form of small data packets.

To support the D-packet service, your ISDN network and financial institution must be equipped with a D-packet handler. To convert the protocol used by the transaction equipment to the X.25 protocol, your ISDN network must also be equipped with an integrated X.25 PAD which works with the following versions of X.25: Datapac 32011, CCITT, T3POS, ITT and API. The ISDN service package you order must include D-packet service (for example, Package P in the United States; Microlink™ with D-channel in Canada).

Your service provider supplies a Terminal Endpoint Identifier (TEI) and DN to support D-packet service. The TEI is a number between 00 and 63 (in Canada, the default range is 21-63). Your service provider may also supply you with a DN to program your D-packet device. The DN for D-packet service becomes part of the dialing string used by the D-packet to call the packet handler.

Service provider features

BCM supports the following ISDN services and features offered by ISDN service providers:

- D-channel packet service (BRI only) to support devices such as transaction terminals. Transaction terminals are used to swipe credit or debit cards and transmit the information to a financial institution in data packets.
- Calling number identification (appears on both BCM sets and ISDN terminal equipment with the capability to show the information).
- Multi-Line hunt or DN hunting which switches a call to another ISDN line if the line usually used by the Network DN is busy. (*BRI only*)

- Subaddressing of terminal equipment (TE) on the same BRI loop. However, terminal equipment which supports sub-addressing is not commonly available in North America. (*BRI only*)

Transmission of B-channel packet data using nailed up trunks is not supported by BCM.

Contact your ISDN service provider for more information about these services and features. For more information about ordering ISDN service in North America, see "Ordering ISDN PRI" on page 711 and "Ordering ISDN BRI" on page 711.

The terminal equipment (TE) connected to the BCM system can use some feature codes supported by the ISDN service provider.

Network name display

This feature allows ISDN to deliver the Name information of the users to those who are involved in a call that is on a public or private network.

Your BCM system displays the name of an incoming call when it is available from the service provider. If the Calling Party Name has the status of *private* it may be displayed as `Private name` if that is how the service provider has indicated that it should be displayed. If the Calling Party Name is unavailable it may be displayed as `Unknown name`.

Your system might display the name of the called party on an outgoing call, if it is provided by your service provider. Your system sends the Business Name concatenated with the set name on an outgoing call but only after the Business Name has been programmed.

The available features include:

- Receiving Connected Name
- Receiving Calling Name
- Receiving Redirected Name
- Sending Connected Name
- Sending Calling Party Name

Consult your customer service representative to determine which of these features is compatible with your service provider.

Name and number blocking (ONN)

(North America only)

When activated **FEATURE 819** allows you to block the outgoing name and/or number on a per-call basis. Name and number blocking can be used with a BCM set.

Consult your customer service representative to determine whether or not this feature is compatible with your provider.

Call by Call Service Selection for PRI

(North America only)

PRI lines can be dynamically allocated to different service types with the Call by Call feature. PRI lines do not have to be pre-allocated to a given service type. Outgoing calls are routed through a dedicated PRI Pool and the calls can be routed based on various schedules.

The service types that may be available, depending on your service provider are described below:

- Public: Public service calls connect your BCM set with a Central Office (CO). DID and DOD calls are supported.
- Private: Private service calls connect your BCM set with a Virtual Private Network. DID and DOD calls are supported. A private dialing plan may be used.
- TIE: TIE services are private incoming and outgoing services that connect Private Branch Exchanges (PBX) such as BCM.
- FX (Foreign Exchange): FX service calls logically connect your BCM telephone to a remote CO. It provides the equivalent of local service at the distant exchange.
- OUTWATS: OUTWATS is for outgoing calls. This allows you to originate calls to telephones in a specific geographical area called a zone or band. Typically a flat monthly fee is charged for this service.
- Inwats: Inwats is a type of long distance service which allows you to receive calls originating within specified areas without a charge to the caller. A toll-free number is assigned to allow for reversed billing.

Consult your customer service representative to determine whether or not this feature is compatible with your provider.

Emergency 911 dialing

(North America only)

The ISDN PRI feature is capable of transmitting the telephone number and internal extension number of a calling station dialing 911 to the Public Switched Telephone Network (PSTN). State and local requirements for support of Emergency 911 dialing service by Customer Premises Equipment vary. Consult your local telecommunications service provider regarding compliance with applicable laws and regulations. For most installations the following configuration rules should be followed, unless local regulations require a modification.

- All PSTN connections must be over PRI.
- In order for all sets to be reached from a Public Safety Answering Position (PSAP), the system must be configured for DID access to all sets. In order to reduce confusion, the dial digits for each set should be configured to correspond to the set extension number.
- The OLI digits for each set should be identical to the DID dialed digits for the set.
- The routing table should route 911 to a PRI line pool.
- If attendant notification is required, the routing table must be set up for all 911 calls to use a dedicated line which has an appearance on the attendant console.
- The actual digit string 911 is not hard-coded into the system. More than one emergency number can be supported.

If transmission of internal extension numbers is not required or desired, then it is recommended that the person in charge of the system maintain a site map or location directory that allows emergency personnel to rapidly locate a BCM set given its DID number. This list should be kept up to date and readily available.

IP telephony note: Ensure that you **do not** apply a 911 route to an IP telephone that is off the premises where the PSAP is connected to the system.

2-way DID

With PRI the same lines can be used for receiving direct inward dialing (DID) and for making direct outward dialing (DOD) calls.

The dialing plan configured by your customer service representative determines how calls are routed. Consult your customer service representative to determine whether or not this feature is compatible with your service provider.

Dialing plan and PRI

The Dialing Plan supports PRI connectivity to public and private networks. The dialing plan is a collection of features responsible for processing and routing incoming and outgoing calls. All PRI calls must go through a dialing plan.

Notes about the dialing plan:

- allows incoming calls to be routed to sets based on service type and digits received
- provides the ability to map user-dialed digits to a service type on a Call by Call basis
- allows long distance carrier selection through user-dialed Carrier Access Codes

Consult your customer service representative to determine how your dialing plan is configured.

ISDN hardware

To support connections to an ISDN network and ISDN terminal equipment, your BCM must be equipped with a BRI S/T Media Bay Module (BRIM) or a Digital Trunk Media Bay Module (DTM) card configured for PRI.

The following describes the hardware:

- “PRI hardware”
- “BRI hardware”

PRI hardware

The Digital Trunk Media Bay Module (DTM) is configured for PRI. In most PRI network configurations, you need one DTM configured as PRI to act as the primary clock reference. The only time when you may not have a DTM designated as the PRI primary clock reference is in a network where your BCM system is connected back-to-back with another switch using a PRI link. If the other switch is loop-timed to your BCM system, your DTM (PRI) can be designated as a timing master.

If your BCM has more than one DTM configured as PRI, you must assign the first DTM as the primary reference, the second DTM as the secondary reference.

If the system has a BRI module, it should be set as the timing master when a DTM in the same network is defined as the primary reference.

BRI hardware

The loops on the BRI module can be programmed to support either network or terminal connections. This allows you to customize your arrangement of lines, voice terminals, data terminals and other ISDN equipment. This section describes some basic hardware configurations for network and terminal connections for each loop type.

A BRI module provides four loops. Each loop can be individually programmed as:

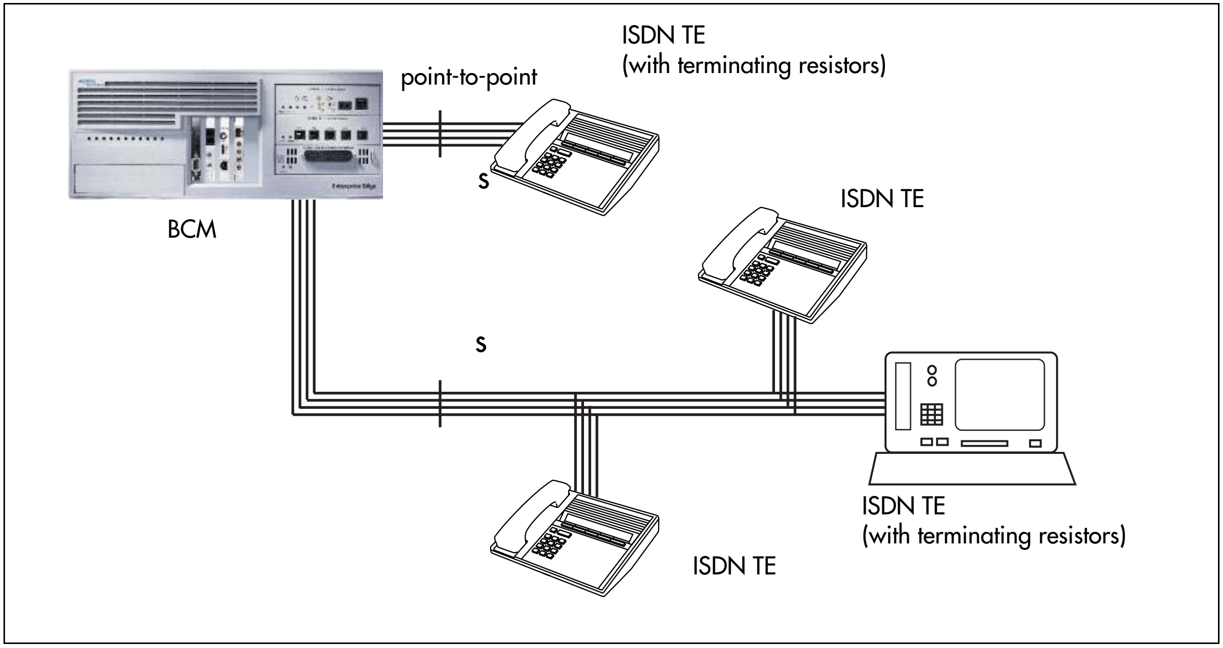
- an S reference point connection (S loop) to ISDN terminal equipment (TE), or
- a T or S reference point connection (T loop or S loop) to an ISDN network using an external NT1

S Reference Point

The S reference point connection provides either a point-to-point or point-to-multipoint digital connection between BCM and ISDN terminal equipment (TE) that uses an S interface. Refer to Figure 220.

S loops support up to seven ISDN DNs, which identify TE to the BCM system.

Figure 220 S reference point

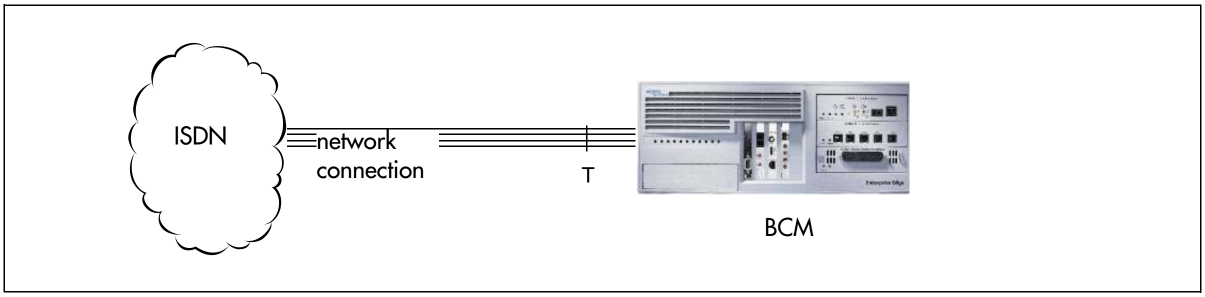


T Reference Points

The T reference point connections provide a point-to-point digital connection between the ISDN network and BCM. Refer to Figure 221.

A T loop provides lines that can be shared by all BCM telephones, peripherals and applications, and ISDN TE.

Figure 221 T reference point



A T loop can be used in combination with an S loop to provide D-packet service for a point-of-sale terminal adapter (POSTA) or other D-packet device. D-packet service is a 16 kbps data transmission service that uses the D-channel of an ISDN line. The T and S loops must be on the same physical module.

Clock source for ISDN

Systems with ISDN interfaces need to synchronize clocking with the ISDN network and any ISDN terminal equipment connected to the network. Systems synchronize clocking to the first functionally available network connection. If there are excessive errors on the reference network connection, the next available network connection is used for clock synchronization. The clock synchronization process generates alarm codes and event messages. Clock synchronization is supported by the DTM, BRI module, and FEM.

The BCM derives timing from the network using T reference points (loops). Terminal equipment on S reference points (loops) derive timing from the BCM system.

When you configure the network connections to the BCM, you should take into account the system preferences for selecting loops for synchronization:

- lower numbered loops have preference over higher numbered loops
- the loop preference order is: 201, 202, 203, 204 etc.
- the system skips S and analog loops, when selecting a network connection for synchronization

Systems with only S loops act as timing masters for the attached terminal equipment (TE), and are not synchronized to the network. ISDN TE without access to a network connection (BRI lines) has limited or no functionality.

If your system has both a BRI S/T configured as BRI, and a DTM configured as PRI, it is recommended that you use PRI as the primary clock source. See “PRI hardware” on page 708.

ISDN BRI NT1 equipment

The NT1 (network termination type 1) connects an S interface (four-wire) to a U interface (two-wire). In most cases, it connects loops from a BRI module to the network connection, which uses the U interface.

The NT1 converts and reformats data so it can be transmitted to and from the S or T connection. In addition, it manages the maintenance messages travelling between the network and the NT1, and between the NT1 and the BCM system.

The NT1 from Nortel is packaged two ways:

- a stand alone package which contains one NT1 card (NTBX80XX) and a power supply (NTBX81XX)
- a modular package which contains up to 12 NT1 cards (NTBX83XX) and a power supply (NTBX86AA)

ISDN standards compatibility

In North America, BCM ISDN equipment supports National ISDN standards for basic call and calling line identification services. BCM BRI is compliant with National ISDN-1 and PRI is compliant with National ISDN-2.

BCM does not support EKTS (Electronic Key Telephone System) on PRI.

In Europe, BCM supports ETSI Euro and ETSI QSIG standards, and PRI SL-1 protocol.

Planning your ISDN network

Consult the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612) to determine a configuration of ISDN trunks and terminal equipment (TE) for the BCM system, then order the appropriate ISDN capability package from your ISDN service provider.

For ISDN BRI service, your service provider supplies service profile identifiers (SPIDs), network directory numbers (Network DNs), terminal endpoint identifiers (TEIs), and other information as required to program your BCM, TE and other ISDN equipment.

BCM does not support any package with EKTS or CACH. EKTS is a package of features provided by the service provider and may include features such as Call Forwarding, Link, Three-Way Calling, and Calling Party Identification.

Ordering ISDN PRI

This section provides information about how to order ISDN PRI service for your BCM.

Ordering ISDN PRI service in Canada

Ordering ISDN PRI service in the Canada/United States from your service provider. Set the BCM equipment to the PRI protocol indicated by your service provider.

Ordering ISDN PRI service outside of Canada and the United States

Outside of Canada and the United States order Euro ISDN PRI and/or BRI service from your service provider. Set the BCM equipment to the Euro ISDN protocol.

Ordering ISDN BRI

The following provides information about how to order ISDN BRI service for your BCM.

Ordering ISDN BRI service in Canada

In Canada, order Microlink™ service, the trade name for standard BRI service. You can order either regular Microlink™ service, which includes the CLID feature, or Centrex Microlink™, which includes access to additional ISDN network features, including Call Forwarding.

When ordering Microlink™ service, it must be ordered with EKTS turned off. If you will be using a point-of-sale terminal adapter (POSTA), ask for D-packet service to be enabled.

Ordering ISDN BRI service in the United States

In the United States, regardless of the CO (Central Office) type, order National ISDN BRI-NI-2 with EKTS (Electronic Key Telephone System) turned off. Use the following packages as a guideline for ordering your National ISDN BRI-NI-2. However, we recommend using packages M or P with the BCM system. Contact your service provider for more information about the capability packages it offers. Bellcore/National ISDN Users Forum (NIUF ISDN packages supported by BCM (for ordering in U.S.).

	Capability	Feature set	Optional features	Point-of-sale	Voice	Data
M	Alternate voice/circuit-switched data on both B-channels	--	CLID	--	X	X
P	Alternate voice/circuit-switched data on both B-channels D-channel packet	flexible calling for voice (not supported by BCM) Basic D-Channel Packet	additional call offering (not supported by BCM) calling line identification	X	X	X

If you want to transmit both voice and data, and support D-channel packet service, order package P. However, BCM does not support the flexible calling for voice and additional call offering features that are included in package P.

Multi-Line Hunt may be ordered with your package. When a telephone number (the Network DN) in the group of numbers assigned by your service providers is busy, the Multi-Line Hunt feature connects the call to another telephone number in the group. BCM supports the feature only on point-to-point, network connections (T loop). Check with your service provider for more information about Multi-Line Hunt.

Any of the ISDN packages will allow you to use sub-addressing, but your ISDN TE must be equipped to use sub-addressing for the feature to work.

Ordering ISDN BRI service outside Canada or the United States

Outside of Canada or the United States order Euro ISDN PRI and/or BRI service from your service provider. Set the BCM equipment to the Euro ISDN protocol.

Supported ISDN protocols

The switch used by your service provider must be running the appropriate protocol software and the correct version of that software to support ISDN PRI and BRI. Each protocol is different and supports different services. Contact your service provider to make sure that your ISDN connection has the protocol you require.

Appendix B

Codec rates

The information in the table below enables the administrator to determine the number of resources that can be maintained on the available system bandwidth.

The packet transfer rate must also include the overhead.

Note: Using Silence Suppression on G.723 and G.729 can reduce the overall bandwidth consumption by 40%.

Note: The totals in the bytes/s column represent one direction only.

Table 181 RTP over IP (Sheet 1 of 2)

Payload (bytes)	Packets/ frame	Overhead (bytes)	Total (bytes)	bytes/s	Overhead (%)	Latency (msec)
G.729						
10	1	58	68	54400	580.00	10
20	2	58	78	31200	290.00	20
***30**	**3**	**58**	**88**	**23467**	**193.33**	**30**
40	4	58	98	19600	145.00	40
50	5	58	108	17280	116.00	50
60	6	58	118	15733	96.67	60
70	7	58	128	14629	82.86	70
80	8	58	138	13800	72.50	80
90	9	58	148	13156	64.44	90
100	10	58	158	12640	58.00	100
G.711						
80	1	58	138	110400	72.50	10
160	2	58	218	87200	36.25	20
***240**	**3**	**58**	**298**	**79467**	**24.17**	**30**
320	4	58	378	75600	18.13	40
400	5	58	458	73280	14.50	50
480	6	58	538	71733	12.08	60
560	7	58	618	70629	10.36	70
640	8	58	698	69800	9.06	80
720	9	58	778	69156	8.06	90

Table 181 RTP over IP (Sheet 2 of 2)

Payload (bytes)	Packets/ frame	Overhead (bytes)	Total (bytes)	bytes/s	Overhead (%)	Latency (msec)
800	10	58	858	68640	7.25	100
G.723						
24	3	58	82	21867	173.33	30
20	3	58	78	20800	160.00	30
Note:*These are the default values.						

Appendix C
Stateful Packet Filters

Overview

BCM supports stateful packet filtering for IP protocols. Stateful packet filters monitor active sessions and record session information such as IP addresses and port numbers. They maintain state information for each flow Stateful filters use the state information to determine if a packet is responding to an earlier request that has been validated by the rule set. If the packet is in response to a previous request, the packet is treated in the same manner. It will either be blocked or allowed through. Stateful packet filters protect networks against Internet attacks such as source spoofing, where an attacker pretends to be a trusted user by using an IP address that is within the accepted range of IP addresses of an internal network. Business Communications Manager 4.0 stateful packet filtering validates that addresses coming from outside the network are valid outside addresses. Stateful packet filters also protect networks from a denial-of-service attack, where an attacker tries to block valid users from accessing a resource or a server.

Understanding Stateful IP Policy Filter Rules

The concept of stateful rules versus non-stateful rules is rather complex. The following describes the IP firewall mechanisms used for the processing of inbound and outbound IP Policy filter rules with the help of flowcharts. In addition, information on stateful session creation and timeout is provided. Finally, to increase the understanding a set of examples is presented to the reader with the help of packet sequence charts.

Figure 222 Processing of Outbound IP Policy Rules

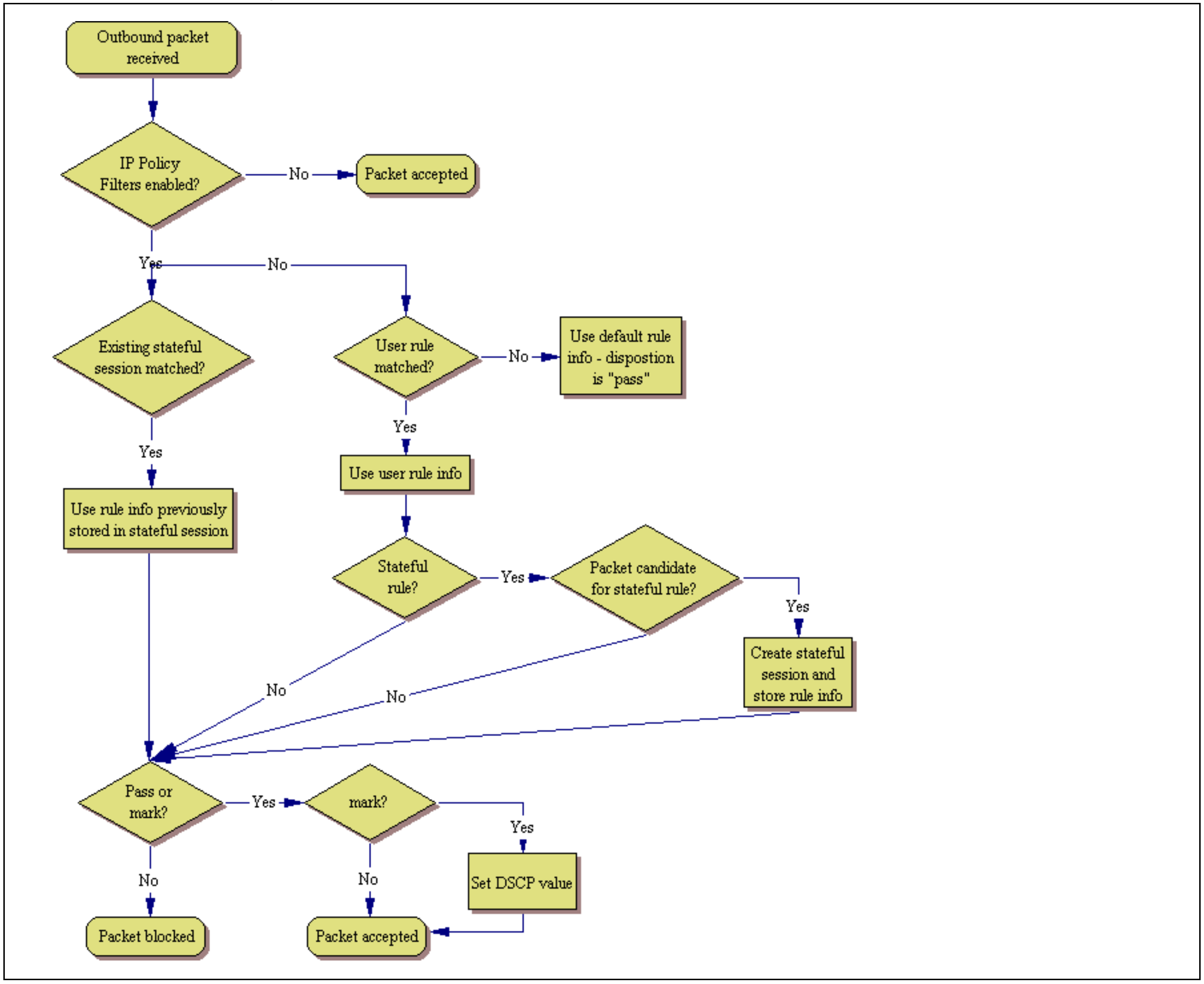
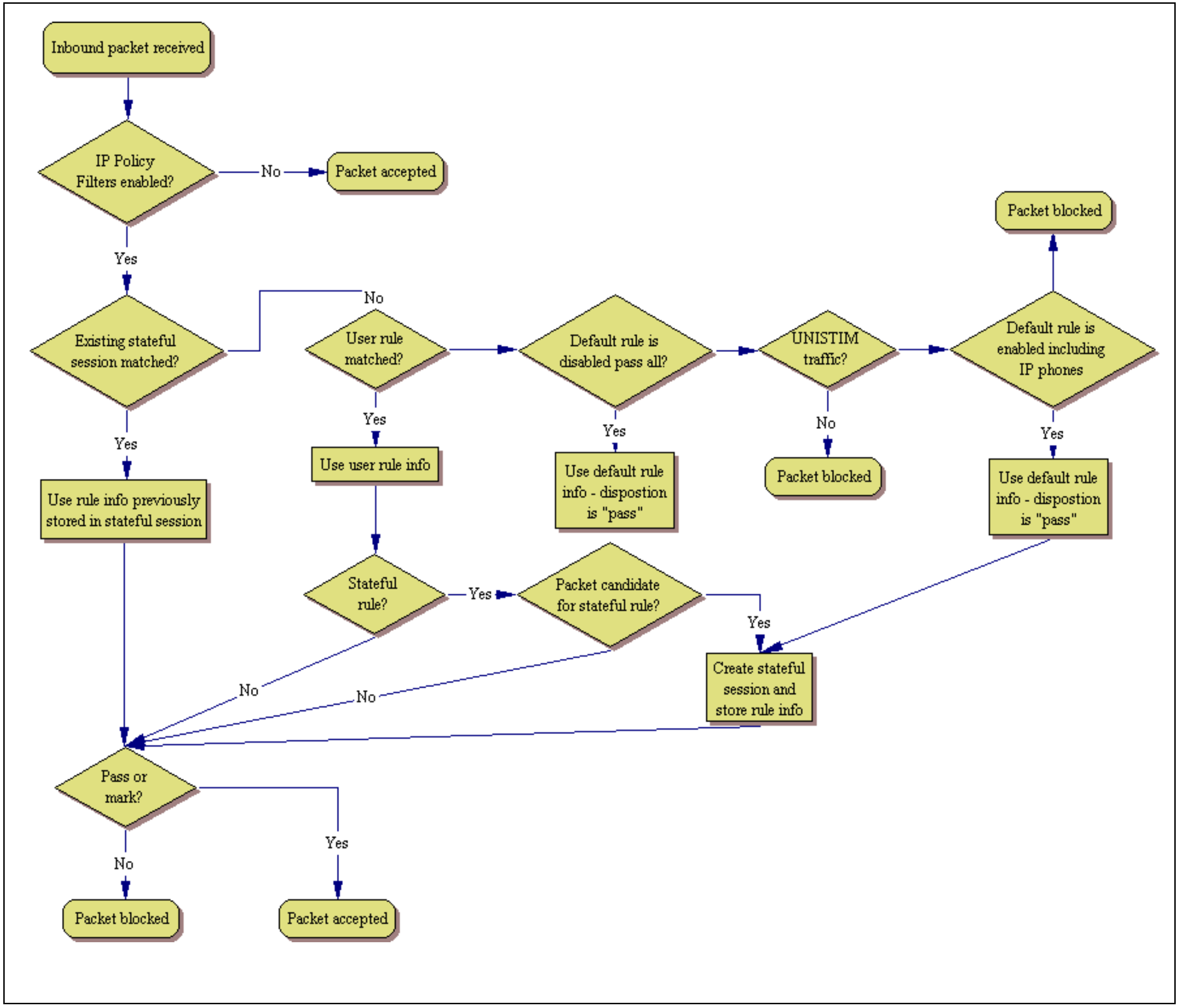

Outbound packet received
IP Policy Filters enabled?
No
Packet accepted
Yes
No
Existing stateful session matched?
User rule matched?
No
Use default rule info - dispostion is "pass"
Yes
Yes
Use user rule info
Use rule info previously stored in stateful session
Stateful rule?
Yes
Packet candidate for stateful rule?
Yes
Create stateful session and store rule info
No
No
Pass or mark?
Yes
mark?
Yes
Set DSCP value
No
No
Packet blocked
Packet accepted

Figure 223 Processing of Inbound IP Policy Rules


Inbound packet received
IP Policy Filters enabled?
No
Packet accepted
Yes
Existing stateful session matched?
No
User rule matched?
Default rule is disabled pass all?
UNISTIM traffic?
Default rule is enabled including IP phones
Packet blocked
Yes
Use rule info previously stored in stateful session
Yes
Use user rule info
Yes
Use default rule info - dispostion is "pass"
No
Packet blocked
Yes
Use default rule info - dispostion is "pass"
Stateful rule?
Yes
Packet candidate for stateful rule?
Yes
Create stateful session and store rule info
No
No
Pass or mark?
No
Yes
Packet blocked
Packet accepted

Stateful session creation

As explained in earlier sections, when the time comes to create a stateful session, the IP header of the packet is inspected. Based on the protocol type and some protocol specific fields, a decision is made to create the stateful session. Once a session is created, it is given an initial timeout value so that it can naturally age out. The session age is refreshed every time a packet is processed for that existing session. Table 182 summarizes the creation of stateful sessions.

To simplify the explanation, the following acronyms are used:

- PT: IP protocol type
- SA: IP source address
- SP: IP source port (applies to UDP and TCP only)
- DA: IP destination address
- DP: IP destination port (applies to UDP and TCP only)

- IID: ICMP protocol session identifier
- ISEQ: ICMP protocol session sequence number

Table 182 Stateful Session Creation

Protocol	Tuple	Timeout	Notes
ICMP	PT, SA, DA, IID, ISEQ	5 secs	The session is selectively created when the following ICMP operations numbers are present the in ICMP playload: 8: echo request 13: timestamp request 15: information request 17: address mask request
TCP	PT, SA, DA, SP, DP	Varies from 30 secs to 2 hours based on TCP state machine	Extensive checks are performed against the TCP state machine after a matching stateful session is retrieved. This ensures early aging of the session in all cases.
UDP	PT, SA, DA, SP, DP	5 mins	The timeout is always 5 minutes except for IKE packets (SP and DP is equal to 500) where the timeout is 8 hours.
UDP	PT, SA, DA, SP, DP	5 mins	The timeout is always 5 minutes except for IKE packets (SP and DP is equal to 500) where the timeout is 8 hours.
ESP, AH	PT, SA, DA	8 hours	
Other protocols	PT, SA, DA	5 mins	

Examples of Stateful Session Processing

Example 1: ICMP

Setup:

Default rule: Enabled – Block incoming except IP phones

Outbound rules: None

Inbound rules:

Disposition: Block, SA: 10.10.10.2/32, stateful is enabled

Assumptions: No stateful sessions present to start

Scenario:

Table 183 Example 1: ICMP (Sheet 1 of 2)

Direction	IP Datagram	Outcome
Outbound	PT: ICMP, SA: 10.10.10.1, DA: 10.10.10.2 ICMP: type 8 (echo request), IID:100, ISEQ: 1	No stateful session [ICMP, 10.10.10.1, 10.10.10.2, 100, 1]] is found. No user rule is found so the default rule is used. A stateful session is created with a disposition to “pass”.

Table 183 Example 1: ICMP (Sheet 2 of 2)

Inbound	PT: ICMP, SA: 10.10.10.2, DA: 10.10.10.1 ICMP: type 8 (echo request), IID:100, ISEQ: 1	Stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 100, 1] is found and the rule is “pass”
	<wait 10 seconds>	Stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 100, 1] is deleted
Outbound	PT: ICMP, SA: 10.10.10.1, DA: 10.10.10.2 ICMP: type 8 (echo request), IID:101, ISEQ: 1	No stateful session [ICMP, 10.10.10.1, 10.10.10.2, 101, 1]] is found. No user rule is found so the default rule is used. A stateful session is created with a disposition to “pass”. “pass”
	<wait 10 seconds>	Stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 101, 1] is deleted
Inbound	PT: ICMP, SA: 10.10.10.2, DA: 10.10.10.1 ICMP: type 8 (echo request), IID:101, ISEQ: 1	No stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 101, 1] is found. An inbound user rule is found and stateful is enabled. A stateful session is created with a disposition to “block”.
Inbound	PT: ICMP, SA: 10.10.10.2, DA: 10.10.10.1 ICMP: type 8 (echo request), IID:101, ISEQ: 1	Stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 101, 1] is found and the rule is “block”

Example 2: UDP with DSCP Marking

Setup:

Default rule: Disabled – Pass all

Outbound rules:

Disposition: Mark with DSCP value 0xC0, SA: 10.10.10.0/24, stateful is enabled

Inbound rules:

1. Disposition: Pass, SA: 10.10.10.3/32, stateful is enabled
2. Disposition: Pass, SA: 10.10.10.4/32, stateful is disabled
3. Disposition: Mark with DSCP value 0xA0, SA: 10.10.10.5/32, stateful is enabled

Assumptions: No stateful sessions present to start

Scenario:

Table 184 Example 2: UDP with DSCP Marking

Direction	IP Datagram	Outcome
Outbound	PT: UDP, SA: 10.10.10.1, SP: 1000, DA: 10.10.10.2, DP: 1001	No stateful session [UDP, 10.10.10.1, 10.10.10.2, 1000, 1001] is found. An outbound user rule is found and stateful is enabled. A new stateful session is created with a disposition to “pass” and “mark” with DSCP value 0xC0.
Inbound	PT: UDP, SA: 10.10.10.2, SP: 1001, DA: 10.10.10.1, DP: 1000	Stateful session with tuple [UDP, 10.10.10.1, 10.10.10.2, 1000, 1001] is found and the rule is “pass” but no marking takes place inbound
Outbound	PT: UDP, SA: 10.10.10.1, SP: 1000, DA: 10.10.10.2, DP: 1001	Stateful session with tuple [UDP, 10.10.10.1, 10.10.10.2, 1000, 1001] found and the rule is “pass” and “mark” with DSCP value 0xC0

Table 184 Example 2: UDP with DSCP Marking

Inbound	PT: UDP, SA: 10.10.10.3, SP: 2001, DA: 10.10.10.1, DP: 2000	No stateful session with [UDP, 10.10.10.1, 10.10.10.3, 2000, 2001] is found. An inbound user rule is found and stateful is enabled. A new stateful session is created with a disposition to “pass” only (but not to mark].
Outbound	PT: UDP, SA: 10.10.10.1, SP: 2000, DA: 10.10.10.3, DP: 2001	Stateful session with tuple [UDP, 10.10.10.1, 10.10.10.3, 2000, 2001] is found and the rule is “pass” only. Hence the packet is not DSC marked.
Inbound	PT: UDP, SA: 10.10.10.4, SP: 3001, DA: 10.10.10.1, DP: 3000	No stateful session with [UDP, 10.10.10.1, 10.10.10.4, 3000, 3001] is found. An inbound user rule is found and stateful is not enabled. No stateful session is created. The packed is accepted.
Outbound	PT: UDP, SA: 10.10.10.1, SP: 3000, DA: 10.10.10.4, DP: 3001	No stateful session [UDP, 10.10.10.1, 10.10.10.4, 3000, 3001] is found. An outbound user rule is found and stateful is enabled. A new stateful session is created with a disposition to “pass” and “mark” with DSCP value 0xC0.

Example 3: UDP and SIP

Setup:

Default rule: Enabled – Block all excluding IP phones

Outbound rules: None

Inbound rules: None

Assumptions: No stateful sessions present to start

Scenario:

Table 185 Example 3: UDP and SIP (Sheet 1 of 2)

Direction	IP Datagram	Outcome
Outbound	PT: UDP, SA: 10.10.10.1, SP: 5060, DA: 10.10.10.2, DP: 5060	No stateful session [UDP, 10.10.10.1, 10.10.10.2, 5060, 5060] is found. No user outbound user rule is found and the default rule is used. A new stateful session is created with a disposition to “pass”.
Inbound	PT: UDP, SA: 10.10.10.2, SP: 5060, DA: 10.10.10.1, DP: 5060	Stateful session with tuple [ICMP, 10.10.10.1, 10.10.10.2, 5060, 5060] is found and the rule is “pass”
	<wait 5 mins>	SIP calls can be made because they were initiated in the outbound direction. However, after 5 minutes of inactivity on the stateful session, it gets deleted.
Inbound	PT: UDP, SA: 10.10.10.2, SP: 5060, DA: 10.10.10.1, DP: 5060	No stateful session with [UDP, 10.10.10.1, 10.10.10.3, 2000, 2001] is found. No user inbound rule is matched, hence the default inbound rule is used with disposition to “block”. No stateful session is created and no SIP call can take place.

Table 185 Example 3: UDP and SIP (Sheet 2 of 2)

	<add an inbound rule as follows: Disposition: Pass, SP: 5060, DP: 5060, stateful is enabled>	This new rule will allow SIP calls to be initiated from the inbound direction.
Inbound	PT: UDP, SA: 10.10.10.2, SP: 5060, DA: 10.10.10.1, DP: 5060	No stateful session with [UDP, 10.10.10.1, 10.10.10.3, 5060, 5060] is found. An inbound user rule is found and stateful is enabled. A new stateful session is created with a disposition to “pass” only (but not to mark]. Now SIP calls can be completed.

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