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BCM 4.0 Telephony Device Installation Guide

BCM 4.0

Business Communications Manager

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Task List

Contents

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 11
- "Related publications" on page 13
- "How to get Help" on page 15

About this guide

The *BCM 4.0 Telephony Device Installation Guide* describes how to configure, and maintain analog, digital, IP, and ISDN devices running on the Business Communications Manager 4.0 (BCM) software.

Purpose

The concepts, operations, and tasks described in this guide relate to the installation and configuration of devices used with the BCM system. This guide provides task-based information on how to configure devices for use with the BCM.

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

In brief, the information in this guide explains:

— installation and configuration of components
— registration and relocation of telephones and devices
— programming loops, configuring digital telephones
— managing system-wide call appearance (SWCA) keys
— setting up central answering positions (CAP)

Audience

The *BCM 4.0 Telephony Device Installation 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

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
- media bay modules (MBM):
    - 4x16
    - ASM8, ASM8+
    - BRIM
    - CTM4, CTM8
    - DDIM
    - DSM16+, DSM32+
    - DTM
    - FEM
    - GASM
    - GATM4, GATM8

BCM features

BCM4.0 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)

BCM applications

BCM4.0 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 13 and User Guides on page 14.

Technical Documents

BCM 4.0 System Overview (N0060607)

System Installation

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

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

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

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)

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)

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)

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

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

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

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
Device description

This chapter describes the telephony devices (telephones) that BCM supports.

Analog devices

BCM supports analog telephones (single-line telephones), cordless telephones, fax machines, answering machines, and modems (with a maximum speed of 28.8 kbit/s). You must install an analog station media bay module (ASM8, ASM8+, and GASM) for analog devices (see Chapter 3, "Installing an analog station media bay module (ASM)). To connect a standard analog voice device or data communication device to the BCM system through a digital station module, you must install an ATA2 (see Chapter 4, "Installing the analog terminal adapter).

Digital devices

BCM supports the following analog devices:

- **T7000**(International only): four memory buttons, without display or indicators
- **T7100**: one-line display, one memory button without indicator
- **T 7208**: one-line display, eight memory buttons with indicators
- **T7316**: two-line display, three display buttons, 16 memory buttons with indicators, eight memory buttons without indicators.

  The T7316 supports separate mute key and a headset key under the dial pad.
- **T7316E**: two-line display, three display buttons, 16 memory buttons with indicators, eight memory buttons without indicators; handsfree, mute, and headset buttons (located under the dial pad)
- **T7406 cordless telephone system:** six memory buttons with indicators and a two-line display with three display buttons.

  The T7406 provides cordless mobility in a small office environment. Each base station supports three telephones. Function is based on the 7316 telephone. The base station connects to a digital station media bay module on the system.
- **Key Indicator Module (KIM)**: 24 memory buttons with indicators
- **BST Doorphone:** used as an intercom to control access to your building. Press the Call button on the BST Doorphone to call one or more telephones, or to send a distinctive chime to telephones in an assigned page zone. Place an internal call from any telephone on the system to the BST Doorphone to set up a two-way voice call. Install a Door Opening Controller to permit the activation of locks on doors or gates.

Wireless devices

BCM supports the following wireless devices:

- **Dect 413x handset:** three display softkeys, four-line handset display, text messaging
- **Dect 414x handset:** three display softkeys, four-line handset display, loudspeaker capability, text messaging
- **Digital Mobility Phone 7420:** three display softkeys, four-line handset display
- **Digital Mobility Phone 7430:** three display softkeys, four-line handset display, text messaging
- **Digital Mobility Phone 7440:** three display softkeys, four-line handset display, loudspeaker capability, text messaging
- **WLAN Handsets 2210/2211/2212:** Voice over IP (VoIP) technology, Push-to-Talk (enables two-way communication with another BCM user)

  The handsets communicate with the BCM system and with the WLAN IP Telephony Manager 2245. Just like wired telephones, the wireless handsets receive calls directly, receive transferred calls, transfer calls to other extensions, and make outside and long-distance calls (subject to corporate restrictions). The handsets interoperates with other IP Line and IP Trunk features and devices, such as IP Peer, and the IP Phone 20xx and IP Softphone 2050 series of IP Phones.

IP devices

BCM supports the following IP devices:

- **IP Phone 2001:** connects through an IP link to the BCM system. The IP Phone 2001 has a single-line text display with a row of display keys on the second display line. The IP Phone 2001 can be used to call through any type of BCM line.
- **IP Phone 2002:** connects through an IP link to the BCM system. The IP Phone 2002 has a two-line text display with a row of display keys on the third display line, and four memory keys with indicators. The IP Phone 2002 can be used to call through any type of BCM line.
- **IP Phone 2004:** connects through an IP link to the BCM system. The IP Phone 2004 has a six-line text display with a row of display keys on the eighth display line, and six memory keys with indicators. The IP Phone 2004 can be used to call through any type of BCM line.
- **IP Phone 2007:** connects to a LAN through an Ethernet connection. The IP Phone 2007 supports call processing features, and can work with an External Application Server to display web-based and interactive applications on the large, color LCD touch screen.
- **IP Softphone 2050:** provides VoIP services using a telephony server and your company’s local area network.
- **IP Audio Conference Phone 2033:** provides audio conferencing. The keypad provides many of the set features of the basic Business Series telephones without display or memory buttons. The audio conference phone comes with three microphones. Installation instructions are provided with the audio conference phone.

- **IP Phone 1120E:** graphical, high-resolution LCD display, backlit, with adjustable contrast. It also has four user-defined feature keys and four soft keys.

  The IP Phone 1120 has a brings voice and data to the desktop by connecting directly to a LAN though an Ethernet connection.

- **IP Phone 1140E:** graphical, high-resolution LCD display, backlit, with adjustable contrast. It also has six user defined feature keys and four soft keys.

  The IP Phone 1140 brings voice and data to the desktop by connecting directly to a LAN through an Ethernet connection.

- **IP Key Expansion Module (KEM):** 24 programmable keys (with labels) for IP Phone 2002 or 2004 models; maximum of four IP KEMs for one phone.

ISDN devices

Refer to "ISDN overview" on page 41 for information on ISDN devices (hardware).

Compatability Matrix

Table 1 is a matrix of telephony devices and the BCM releases with which they are compatible. Table 1 also shows what media bay module (MBM) is needed to support each device.

Table 1 Telephony devices release compatibility matrix

Device	BCM 3.5	BCM 3.6	BCM 3.7	BCM50 1.0	BCM 4.0	MBM
T7100	X	X	X	X	X	DTM
T7100	X	X	X	X	X	DTM
T7208	X	X	X	X	X	DTM
T7316	X	X	X	X	X	DTM
T7316E	X	X	X	X	X	DTM
T7406	X	X	X	X	X	DTM
T 24 KIM	X	X	X	X	X	DTM
BST Doorphone	X	X	X		X	DTM
Dect 413x			X		X	DSM
Dect 414x			X		X	DSM
Digital Mobility Phone 7420					X	DTM
Digital Mobility Phone 7430			X		X	DTM
Digital Mobility Phone 7440			X		X	DTM
IP Phone 2001		X	X	X	X	
IP Phone 2002	X	X	X	X	X	
IP Phone 2004	X	X	X	X	X	

Table 1 Telephony devices release compatibility matrix

Device	BCM 3.5	BCM 3.6	BCM 3.7	BCM50 1.0	BCM 4.0	MBM
IP Phone 2007			X		X	
IP Phone 1110					X	
IP Phone 1120E					X	
IP Phone 1140E					X	
IP Softphone 2050	X	X	X	X	X	
IP Audio Conference Phone 2033					X	
IP KEM					X	
WLAN 2210 Handset			X		X	
WLAN 2211 Handset			X		X	
WLAN 2212 Handset					X	

Table 2 shows the types of lines supported by different MBMs and the number of lines those MBMs support.

Table 2 MBM trunk requirements

Type of lines	Type of MBM	Number of lines per MBM
T1 digital	digital trunk MBM (DTM)	24
Universal T1 MUX digital lines	digital drop and insert MUX (DDIM)	24 (also requires a full DS30 channel for the data module)
PRI digital lines (NA)	DTM	23
E1 digital lines	DTM	30
PRI digital lines (EMEA)	DTM	30
Analog lines	caller ID trunk module 4(CTM4) (North American systems only)	4
Analog lines	CTM8 (North American systems only)	8
Analog lines	global analog trunk module 4 (GATM4)	4
Analog lines	GATM8	8
Analog lines	4x16 combination MBM (North American systems only)	4 (also requires a full DS30 channel for the DNs)
BRI ISDN lines	BRIM S/T	4 ISDN loops

Table 3 MBM station requirements

Type of extension	Type of MBM	Number of extensions per MBM
Digital extensions	DSM16/DSM16+	16
Digital extensions	DSM32/DSM32+	32
Digital extensions	4x16	16
Analog extensions	ASM8	8
Analog extensions	GASM8	8
Cordless handsets (DECT) (selected profiles only)	DSM	32
Digital extensions are for digital or IP telephones. You do not need to include IP telephones when you calculate the number of required DSM MBMs.		

Chapter 3
Installing an analog station media bay module (ASM)

The analog station media bay modules (ASM8, ASM8+, and GASM) can connect to a maximum of eight analog telecommunication devices. These devices are standard analog telephones, cordless telephones, fax machines, answering machines, or modems. The maximum speed for a modem connection is 28.8 kbit/s.

The ASM8 is available in North America only; the ASM8+ and GASM8 are available in North America, the United Kingdom, Australia, and Poland.

In addition to ASM8 features, the ASM8+ and GASM offer the following features:

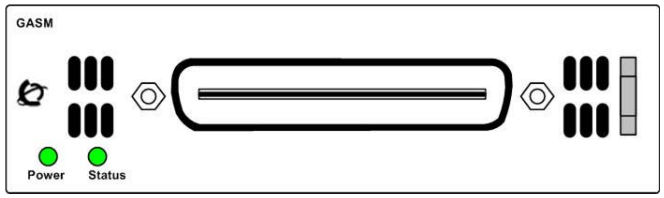
- Visual Message Waiting Indicator (VMWI) — LED indicates to the end user that a message is waiting.
- Disconnect supervision (Open Switch Interval [OSI] as per EIA/TIA 464) — indicates to the attached device, in an established communication, that the connected device should release the call (see disconnect supervision note).
- Caller ID — provides the name, phone number, and other information about the caller to the end user at the start of the call.
- Firmware downloading capability — allows the system to upgrade the ASM8+ and GASM firmware at customer sites.
- Enhanced ringing capability — ASM8+ and GASM provide a ringing voltage of 2 REN/65 V rms per port.
- Calling line identification (CLID)
- The GASM8 is designated as an ONS (on-premise station) port.

Disconnect supervision note: When disconnect happens from the central office, the ASM8+ provides an open switch interval (OSI) to the off-hook station of 850 ms (TIA/EIA 464 section 5.4.10.2.4; minimum is 600 ms) as a disconnect signal. If the station remains on-hook after the disconnect signal, the ASM8+ disconnects the station equipment from the network without returning a tone to it (TIA/EIA 464 section 5.4.10.2.5[1]). After the station equipment goes on-hook, the ASM8+ station interface is restored to on-hook (idle).

It is important to ensure that the device, application, or interface card connected to an ASM8+ station interface conform to these on-hook and off-hook conditions.

The ASM8, ASM8+, and GASM each have one RJ-21 connector on the faceplate. Figure 26 shows the GASM.

Figure 1 GASM faceplate LEDs and connectors



The ringer equivalency number (REN) per port for ASM8 is 1; the REN for ASM8+ and GASM is 2.

Note: The termination of the analog interface can consist of any combination of devices, subject only to the requirement that the sum of the RENs of all the devices does not exceed the REN of the interface to which the device is connected.

Refer to the following for information on installing and configuring an ASM:

- "Installing an MBM"
- "Configuring the media bay module" on page 66
- "Wiring the ASM" on page 67
- "Installing analog devices" on page 68

For more detailed information on installing the BCM system and related components, refer to *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

Installing an MBM

MBMs are installed in BCM main units and expansion units, depending on your system requirements.

The primary tasks to install an MBM are:

- Selecting MBMs for your system
- Assigning DS30 resources
- Setting MBM dip switches
- Installing an MBM

For more detailed information on installing an MBM, refer to *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

Configuring the media bay module

For information on installing a media bay module (MBM) and setting the dip switches, refer to the *BCM200/400 4.0 Installation and Maintenance Guide* (N0060612).

To configure the MBM

1 Open Element Manager and connect to your BCM system.

2 Click **Configuration > Resources > Telephony Resources**.
The **Telephony Resources** panel appears (see Figure 27).

3 In the **Modules** table, select the location of the MBM that you want to configure.

4 Double-click the **Programmed type** field to display the drop-down list.

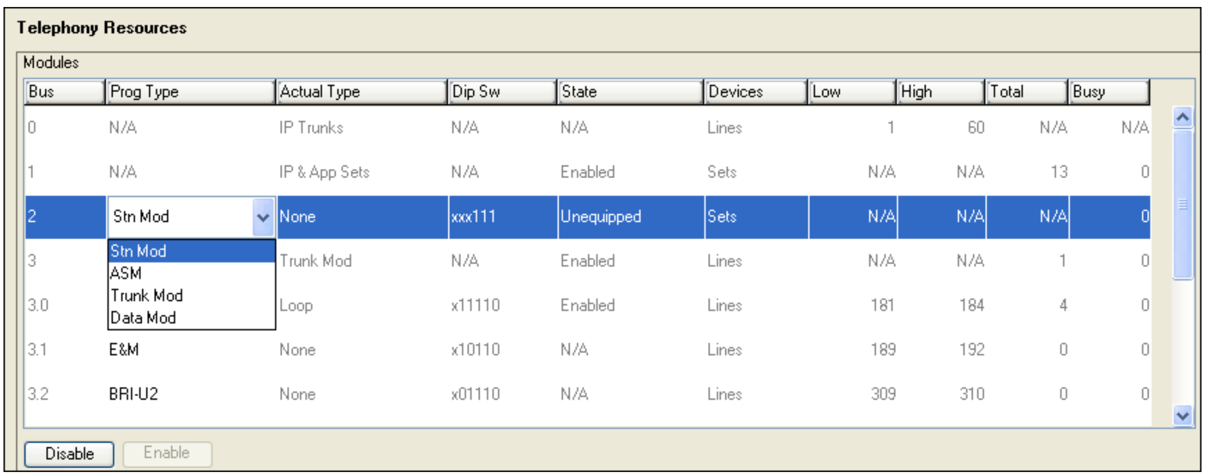
5 Select the type of MBM that you installed in that location.

6 Click **Enable**.

7 Repeat steps 4 to 7 to enable each MBM in your system.

You can set other parameters for the MBMs depending on the type of MBM you installed.

Figure 2 Telephony Resources panel



Wiring the ASM

An experienced installer can wire the ASM for your system using the wiring chart, for more information refer to the "ASM8, ASM8+, and GASM wiring chart" on page 83.

Installing analog devices

After the ASM is correctly wired, you can connect your analog devices.

Documentation describing installation and features of your analog devices is supplied with each piece of equipment.

Chapter 4
Installing the analog terminal adapter

The following provides installation instructions for the analog terminal adapter 2 (ATA2) or ATA.

The ATA2 connects a standard analog voice device or data communication device to the BCM system through a digital station module. Examples of analog voice devices are analog telephones and answering machines. Examples of analog data communication devices are modems and fax machines.

The ATA2 is designated as either an ONS (on-premise station) or an OPS (off-premise station) port.

Refer to the following topics for information on installing an ATA2:

- "Configuration overview"
- "Installing the ATA2" on page 28
- "Configuring the ATA2" on page 31

Configuration overview

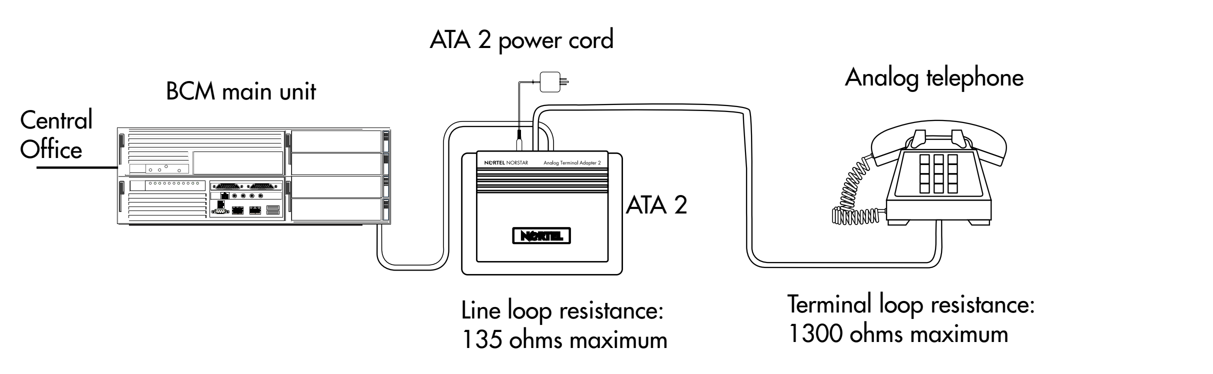
The following describes environment configurations for connecting analog and data devices to the main unit using an ATA2:

- "Analog telephone"
- "Analog data device" on page 28

Analog telephone

Figure 3 on page 27 shows an installation overview for connecting an analog device through an ATA2 to the main unit.

Figure 3 Analog telephone installation overview

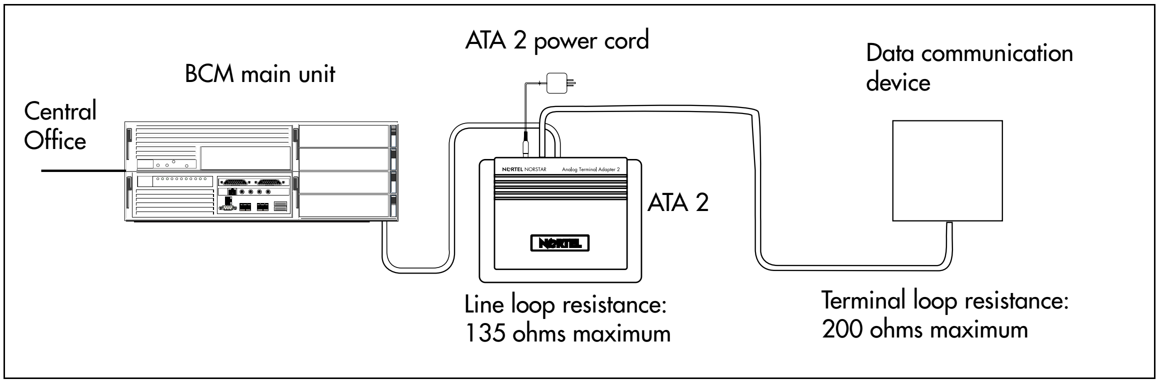


Analog data device

The ATA2 connects a standard analog data device, such as a fax or modem, to the BCM system.

Figure 4 shows an installation overview for connecting a data communication device through an ATA2 to the BCM system.

Figure 4 Data communication device installation overview



Installing the ATA2

The following provides information on installing the ATA2:

- "Connecting the ATA2"
- "Mounting the ATA2" on page 29
- "Test insertion loss measurement" on page 30

Connecting the ATA2

After the correct environment has been set up, connect the BCM system and the analog device to the ATA2 and then connect the power (see Figure 5).

Figure 5 ATA2 top view

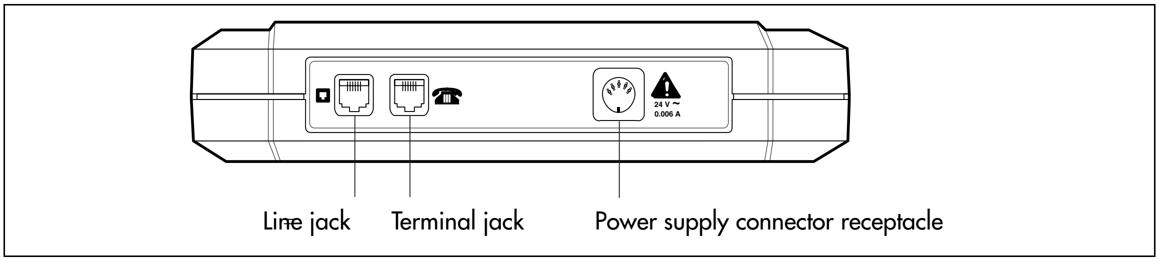
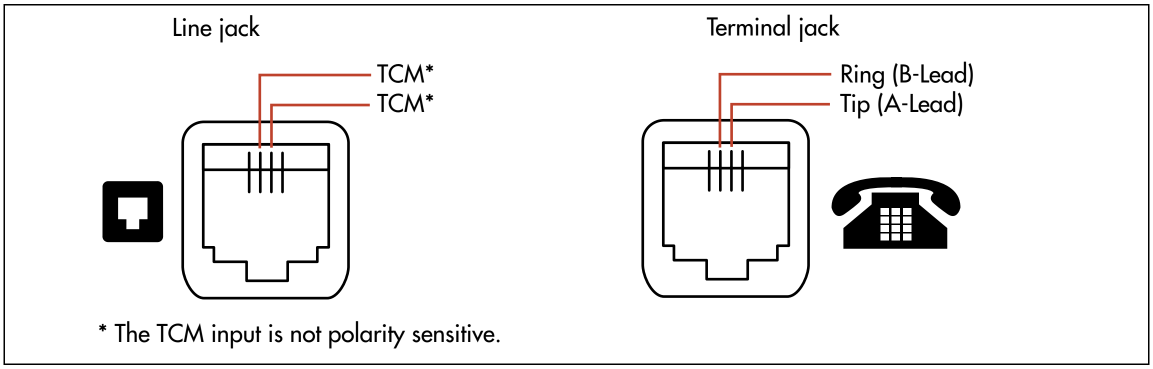


Figure 6 shows the pin-outs for the connection cables.

Figure 6 ATA2 pin-outs



To connect the ATA2

1. Connect one end of a line cord to the ATA2 terminal jack.
2. Connect the other end to your telephone, modem, or fax machine.
3. Connect one end of a line cord to the ATA2 line jack.
4. Connect the other end to an available station port on the BCM main unit or expansion unit.
5. For a 120 V or 230 V system, plug the DIN connector of the power supply cord into the power supply connector receptacle. Plug the adapter into a standard AC outlet.

Caution: In North America, the ATA2 must be powered from a Class 2 power source that is UL- and CSA-approved.

In Europe, the ATA2 must be powered from a Class II power source that is CE marked.

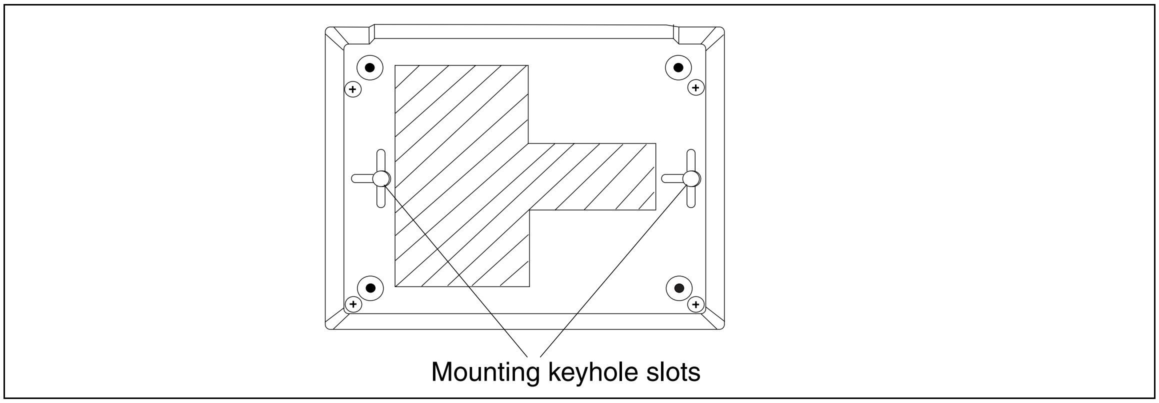
Mounting the ATA2

After the ATA2 is correctly connected, you can mount the unit on a wall, as described in this section.

To mount the ATA2 on a wall

1. When using 0.5 mm wire (24 AWG), select a location within 800 m (2600 ft.) of the BCM main unit.
2. Allow 12.5 cm (5 in.) clearance for the line jack, terminal jack, and power supply connector.
3. Screw two 4-mm (#8) screws into the wall, 130 mm (5 1/4 in.) away from each other. Leave 6 mm (1/4 in.) of the two screws showing.
4. Align the slots at the back of the ATA2 unit over the screws. Push the unit against the wall. The line jack, terminal jack, and power supply connector must be at the top of the ATA2 (see Figure 7).

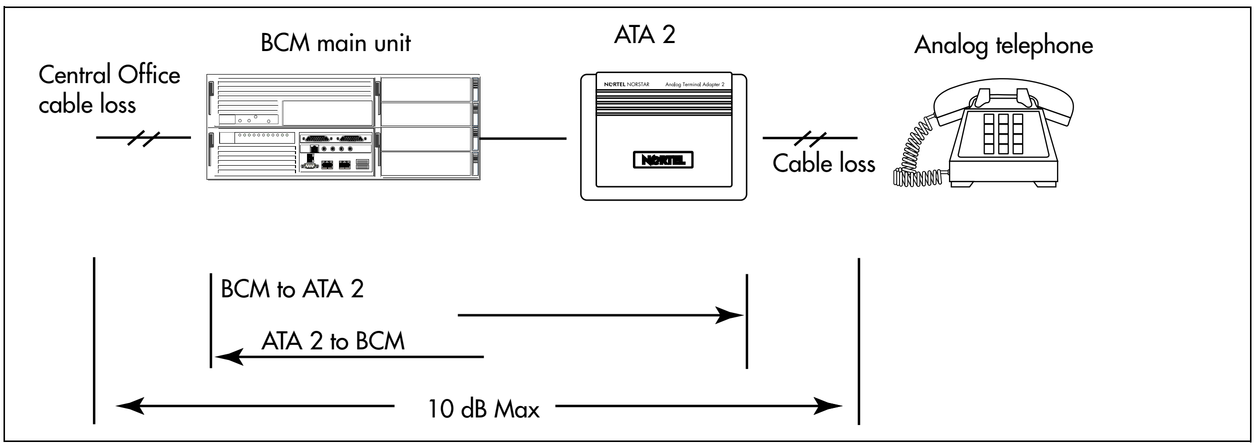
Figure 7 ATA2 back view



Test insertion loss measurement

The maximum loss for ATA2 to Central Office (CO) configuration must not exceed 10 dB (see Figure 8 on page 30).

Figure 8 Insertion loss from the CO to the analog telephone



Longitudinal balance to ground	50 dB 60 to 4,000 Hz With IEEE 455-1976 test
Overload level	3 dB

Measure the total insertion loss between the CO and analog device by using standard dial-up test lines with a transmission test set (for example, Hewlett-Packard 4935A Transmission Test Set).

To measure the insertion loss from the CO to the analog device

1 Establish a connection to the 1 mW, 1 kHz, CO service line with an analog telephone attached to the ATA2.

2 Ensure that the analog port terminates correctly in 600 ohms:

- Replace the analog telephone with the test set.

- Use RECEIVE/600 OHM/HOLD mode on the test set.

3 Ensure that the test set connects in parallel to the service line before removing the analog telephone or the line drops.

4 Remove the single-line telephone.

5 Measure the 1 kHz tone at the far end of the analog port, which is where the analog loop ends and where the analog device connects.

Note: The tone must be greater than - 10 dB (for example, - 9 dB is acceptable).

To measure the insertion loss from the analog device to the CO

1 Establish a connection to a silent termination on the CO service line with an analog telephone attached to the ATA2.

2 Make sure the analog port terminates correctly in 600 ohms:

- Replace the analog telephone with the test set.
- Use TRANSMIT/600 OHM/HOLD mode on the test set.

3 Make sure the test set connects in parallel to the service line before removing the analog telephone or the line drops.

4 Remove the analog telephone.

5 Introduce a 1 kHz tone into the analog line at - 10 dBm, and measure the level at the CO exchange.

Note: The difference in levels is the transmit loss and must be less than 10 dB (for example, 9 dB is acceptable).

Configuring the ATA2

Configure the ATA2 using Element Manager or Telset Administration. For detailed configuration information, refer to the *BCM 4.0 Device Configuration Guide* (N0060600).

Chapter 5
Using an analog telephone

The following explains how to make, and answer, calls and how to access features on analog telephones. Features described in this guide are for analog telephones with a **LINK** or **FLASH** button.

Note: Analog telephones in Europe or Australia have a **RECALL** button instead of a **LINK** or **FLASH** button.

If your telephone does not have a **LINK**, **FLASH**, or **RECALL** button, you must use the Hook Switch (located under the handset). The Hook Switch must be pressed for approximately one half of one second.

If your telephone does not have a [*] or [#] button, you must use dialpad numbers. To indicate a [*], enter the number [1]. To indicate a [#], enter the number [3].

Making and answering calls

Refer to the following procedures to make and answer calls:

- "To make external calls"
- "To make internal calls"
- "To answer calls" on page 34
- "To make or answer a second call" on page 34
- "To answer a second call while on another call" on page 34
- "To hold a call and make a second call" on page 34

To make external calls

1 Lift the handset.

2 Dial the external code (or line pool code) to access an external line.

3 Dial the telephone number.

Contact your system administrator to confirm what external code or line pool code to use on your telephone.

To make internal calls

1 Lift the handset.

2 Dial the extension number.

Contact your system administrator for a list of extension numbers.

To answer calls

Lift the handset to answer a call when your telephone rings.

To make or answer a second call

You can have two calls active at the same time. Use **LINK** 2 to switch between calls.

To answer a second call while on another call

Press **LINK** 2 to answer the second call. The first call is automatically placed on hold.

To hold a call and make a second call

Press **LINK** 2 to place the first call on hold. Dial the telephone number of the second call.

Call Display Information

Depending on the system hardware/software configuration, Call Display information (CLID) for incoming external calls can be viewed on analog display telephones. Your system administrator must enable the CLID feature for your telephone in system programming.

The name and number of an external caller appears on the telephone display after the first ring (second ring if this is an analog line).

Note: Not all analog display telephones are capable of showing name and number information.

Contact your system administrator for more information on Call Display capabilities for your telephone.

Message Waiting Indication (MWI)

Depending on the system hardware/software configuration, visual or audible signalling for Message Waiting Indication (MWI) is available for analog telephones.

There are two MWI types: Stutter Dial Tone and Lamp Indication. Your system administrator determines which MWI type is assigned to your telephone in system programming:

- **Stutter Dial Tone**
  Lift the handset. You hear a stuttered dial tone when you have a message waiting.
- **Lamp Indication**
  The indictor lamp on your telephone lights when you have a message waiting. The lamp indication supported on the GASI ports are low voltage, and do not meet the typical CO voltage requirements.

To cancel MWI

Press **LINK** # 6 5 or reply (listen) to your new messages. Since MWI is only active when you have a new message, after you have replied (listened) to your message, it is no longer a new message, and MWI is canceled.

Replying to messages

You can receive internal and external messages.

- Internal messages are sent from a designated direct-dial telephone or an internal voice message center on your system.
- External messages are sent from a remote voice message center outside your system.

Contact your system administrator to confirm your mailbox privileges on an internal or remote voice message center.

To reply to internal messages

1. Press **LINK** * 6 5 to be automatically connected to the internal message sender. If you have more than one message waiting, you are connected to the sender of the first received message.
2. Dial the extension for the internal voice message center. Enter your mailbox number and password and press #. Follow the voice prompts to access your messages. Contact your system administrator for the extension number of the internal voice message center.
3. For more information on internal voice messaging features, refer to "Voice Messaging - Internal" on page 38.
4. Dial the single-digit access code for the designated direct-dial telephone to retrieve your messages. Contact your system administrator for the single-digit access code.

To reply to external messages

Place a call to the remote voice message center to retrieve your messages.
Contact your system administrator for the telephone number of the remote voice message center.

Feature list

Table 4 lists available telephone features.

Table 4 Telephone features and descriptions (Sheet 1 of 3)

Feature	Description
Call Forward	**LINK** * 4 **Cancel: LINK** # 4 Directs your calls to another telephone connected to your system. Press **LINK** * 4 followed by the extension number of the telephone that is to receive the forwarded calls.
Call Park	**LINK** * 7 4 Parks the call on hold and allows it to be retrieved from any other telephone within the system. When the call park is successful, you hear a confirmation tone, and the call is parked on the highest numbered park code in the system. If call park is unsuccessful, you hear an error tone, and remain connected with the call. **To retrieve a parked call**: Lift the handset, and dial the retrieval code. Contact your system administrator for a list of park codes. For analog devices, Call Park is activated on the last Call Park port (for example, X25).
Call Pickup, directed	**LINK** * 7 6 and the extension number of the ringing telephone. Allows you to answer any ringing telephone in your system.
Call Pickup, group	**LINK** * 7 5 Allows you to answer any ringing telephone within your pickup group.
Call Queuing	**LINK** * 8 0 1 Allows you to answer the next call. If more than one call is waiting, priority is given to incoming external calls over callback, camped, or transferred calls.
Camp-on	**LINK** * 8 2 and the extension number Allows you to reroute a call to another telephone even if all the telephones’ lines are busy.

Table 4 Telephone features and descriptions (Sheet 2 of 3)

Feature	Description
Conference	**LINK** * 3 Allows you to establish a three-way conference among yourself, one external call, and one internal call, or yourself and two internal calls. Line pool access allows you to establish a conference with yourself and two external calls.
	To establish a conference: Make or answer the first call. Press **LINK** 2. The first call is automatically placed on hold. Make or answer the second call. Press **LINK** * 3 to complete the conference. If the second call is busy, replace the handset, and **LINK** 2 to return to the first call.
	To put a conference on hold: Press **LINK** 2.The other two callers can still talk to each other. **To return to the conference call**: Press **LINK** 2 **again**.
	To split a conference: Press **LINK** # 3. This allows you to place one caller on hold, and to consult with the other caller. Press **LINK** 2 to alternate between callers. **To reestablish the conference**: Press **LINK** * 3.
	To disconnect one party: Press **LINK** # 3. This allows you to place one caller on hold. Press **LINK** 2 to alternate between callers. **To end a call:** Finish your conversion then replace the handset. **To retrieve the held call:** Press **LINK** 2.
Hold Call - Exclusive	**LINK** * 7 9 Allows you to place an active call on hold, and prevents the held call from being picked up from other telephones.
Hold Call - Public	**LINK** 2 Allows you to place an active call on hold, and allows the held call to be picked up from other telephones.
Last Number Redial	**LINK** * 5 Automatically dials the last external telephone number you dialed.
Page	Contact your system administrator for a list of page zones. Internal page: **LINK** * 6 1 **and zone (0 to 6)** Make a page announcement to all telephones, or to a specific group of telephones, through the telephone speakers. Zone 0 pages all zones.
	External page: **LINK** * 6 2 Make a page announcement through an external loudspeaker system.
	Internal and external page: **LINK** * 6 3 **and zone (0 to 6)** Make a page announcement through both your telephone speakers and an external loudspeaker system. Zone 0 pages all zones.

Table 4 Telephone features and descriptions (Sheet 3 of 3)

Feature	Description
Privacy Control	**LINK** * 8 3 Prevents another telephone that shares your line from joining your current call, or permits another telephone that shares your line to join your active call.
Ring Again	**LINK** * 2 **Cancel: LINK** # 2 Allows you to monitor a busy or unanswered telephone, or a busy line pool within your system. When the telephone or line pool becomes available, you are alerted with one short ring from your telephone.
Saved Number Redial	**LINK** * 6 7 Allows you to save an external number to redial later. Enter **LINK** * 6 7, while you are on a call, to save the external number. **To automatically redial the saved number:** Lift the handset and press **LINK** * 6 7.
Send Message	**LINK** * 1 **Cancel: LINK** # 1 Allows you to send a message to another telephone within your system.
Transfer	**LINK** * 7 0. Allows you to send a call to another telephone within your system. To transfer a call, you must have an available internal line.
	Transferring a call: Make or answer a call. Press **LINK** * 7 0. Dial the extension number of the telephone to receive the transfer call. Replace the handset to complete the transfer.
	Transfer using announce: Make or answer a call. Press **LINK** * 7 0. Dial the extension number of the telephone to which you want to transfer the call. Announce the call to the receiving party. Replace the handset to complete the transfer.
	Transfer using conference: Make or answer a call. Press **LINK** 2. The call is put on hold temporarily. Make a second call. Press **LINK** * 3 Replace the handset to complete the transfer.
Trunk Answer	**LINK** * 8 0 0 Allows you to pick up a ringing external call on a line placed into a Ringing Service schedule.
Voice Call	**LINK** * 6 6 and the extension number Allows you to make a voice announcement, or begin a conversation, through the speaker on a digital telephone, without making the other telephone ring.
Voice Messaging - Internal	**LINK** * 9 8 1 and follow the voice prompts. Allows you to open your mailbox to listen to your messages.
	LINK * 9 8 0 and follow the voice prompts Allows you to record and send a message directly to a mailbox, without calling the extension.
	LINK 4 Allows you to forward calls directly to your mailbox.

Other documents

Refer to the *BCM 4.0 Telephone Features User Guide* (N0027160) for a complete list of features available for all types of telephones on your system.

Note: You press the **FEATURE** button on digital telephones to access features. You press **LINK** [*], **FLASH** [*], or **RECALL** [*] buttons on analog telephones to access features.

Chapter 6
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 43
- "ISDN hardware" on page 47
- "ISDN standards compatibility" on page 50
- "Planning your ISDN network" on page 50
- "Supported ISDN protocols" on page 52

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 42
- "ISDN layers" on page 42
- "ISDN bearer capability" on page 43

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.

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 45
- “Name and number blocking (ONN)” on page 45
- “Call by Call Service Selection for PRI” on page 45
- “Emergency 911 dialing” on page 46
- “2-way DID” on page 47
- “Dialing plan and PRI” on page 47

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 50 and "Ordering ISDN BRI" on page 50.

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:

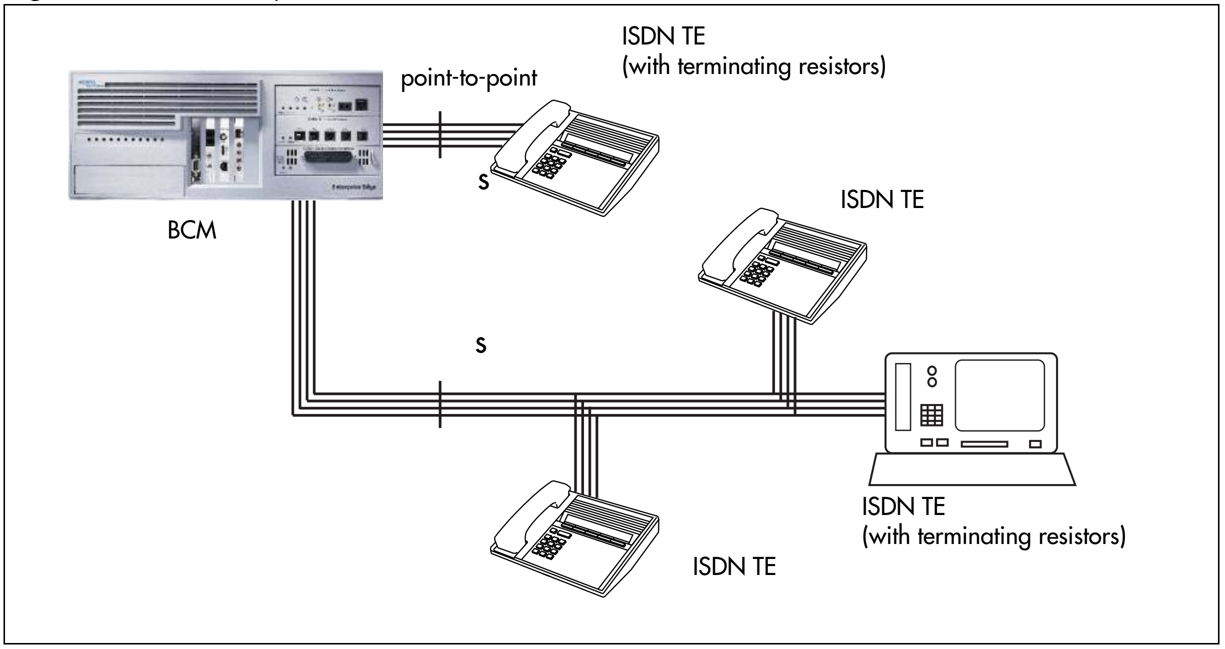
- an S reference point connection (S loop) to ISDN terminal equipment (TE), or
- a T or S reference point connection (T loop or S loop) to an ISDN network using an external NT1

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 9.

S loops support up to seven ISDN DNs, which identify TE to the BCM system.

Figure 9 S reference point

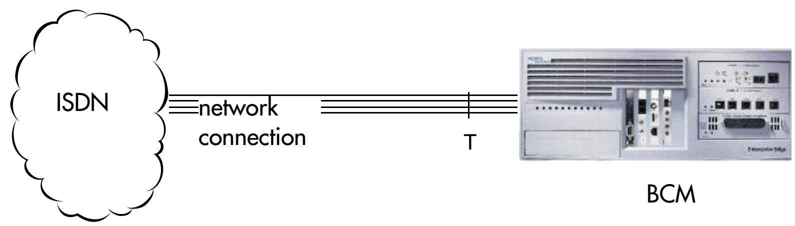


T Reference Points

The T reference point connections provide a point-to-point digital connection between the ISDN network and BCM. Refer to Figure 10.

A T loop provides lines that can be shared by all BCM telephones, peripherals and applications, and ISDN TE.

Figure 10 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 47.

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

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.

Chapter 7
Telephone button icons

The digital phone Feature button is a small globe icon. The legacy digital phone Feature button reads Feature or Fx. The IP telephones display Feature above the far left display key, when feature selection is available.

The appearance of **FEATURE** indicates pressing the Feature key before entering a feature code. The table below shows which buttons to use on the different types of Nortel telephones to use the features. Refer to each user card for specific details about each type of telephone.

Button Function	Business Series Terminals (T-series)	Legacy telephones (M-series)	IP telephones
Feature		Feature, Fx	Display key
Hold		Hold, 	
Release		Rls, 	
Answer call	Telephones with line buttons: Press active line button or Intercom key and lift handset. Telephones with no buttons: Lift handset.		

The following labels are used to indicate each type of configuration button:

- **FEATURE** indicates pressing the Feature key.
- **HOLD** indicates pressing the Hold key.
- **RLS** indicates pressing the Release key.

Note: Your telephone may not have access to all the features listed in this guide, either because your telephone does not support the feature, or because the feature has not been enabled at your telephone. Your system administrator can provide details.

Telephone features

Table 5 Telephone features (Sheet 1 of 7)

Feature	Description
Background Music	**FEATURE 86** **Cancel: FEATURE #86** Listen to music (provided by an external source or an IP source connected to the system) through your telephone speaker when you are not on a call.
Button Inquiry	**FEATURE *0** Check what is programmed on any button. Use when labeling buttons.
Call Duration Timer	**FEATURE 77** Briefly display the approximate length of your current or most recent call.
Call Forward	**FEATURE 4** **Cancel: FEATURE #4** Send your calls to another telephone in your system.
Call Park	**FEATURE 74** Put a call on hold so that it can be picked up from any telephone in your system. The display shows a three-digit retrieval code. To retrieve a parked call: press an intercom button and dial the retrieval code. On model 7000, 7100, and 2001 telephones, lift the handset and dial the retrieval code.
Call Pickup, directed	**FEATURE 76** and the telephone number. Answer any ringing telephone.
Call Pickup, group	**FEATURE 75** Answer a call that is ringing at another telephone in your pickup group. The external call that has been ringing longest is answered first.
Call Queuing	**FEATURE 801** Answer the next call. If more than one call is waiting, priority is given to incoming external calls over callback, camped, or transferred calls.
Camp-on	**FEATURE 82** and the extension number of the receiving telephone Re-route a call to another telephone even if all lines on that telephone are busy.
Class of Service Password	**FEATURE 68** plus CoS password Change the dialing filters on a line or telephone, or gain external access to your system. Dialing filters determine which numbers you can dial. The CoS password is provided by your system administrator to change your class of service.
Conference	**FEATURE 3** **Establish a conference call between yourself and two other parties.** 1. Place or answer the first call. 2. Put the first call on hold. 3. Place or answer the second call. 4. After the second call is connected, press **FEATURE 3**. 5. Press the line or intercom button of the first held call (not required on model 7000, 7100, or 2001 telephones). 6. Press **RLS** to end the conference call.

Table 5 Telephone features (Sheet 2 of 7)

Feature	Description
	To remove yourself from a conference permanently (unsupervised conference): Press **FEATURE 70**. The other two callers remain connected. (Some external lines do not support this feature.)
	To put a conference on hold: Press **HOLD**. The other two callers can continue to talk to each other.
	To split a conference: Press the line or intercom button of one caller to consult privately while the other caller is on hold. **To re-establish the conference:** Press **FEATURE 3.**
	To disconnect one party: 1. Press the line or intercom button for the caller you want to disconnect. 2. Press **RLS**. 3. Press the line or intercom button for the remaining caller to resume your conversation.
	To independently hold two calls: 1. Press the line or intercom button of the first caller. 2. Press **HOLD**. The second caller is put on hold automatically. **To re-establish the conference:** 1. Retrieve one call from hold. 2. Press **FEATURE 3**. 3. Retrieve the second call from hold.
	To send Hookswitch or DTMF during a conference call Either system telephone engaged in a three-way conference call over a Network CLID or DS trunk can issue a hookswitch or DTMF dialing request without leaving the conference, if the feature is enabled. **Note:** This feature is not available for IP telephones. – To hear DTMF tones on both telephones during dial, activate Long Tones (**FEATURE 808**). – To conference in someone through the trunk, use **Link** (**FEATURE 71**)
Contrast adjustment	**FEATURE *7** plus a number from 1 to 9 to adjust the display contrast. Press **HOLD** to set your choice.
Dialing modes	**FEATURE *82** Choose one of three methods of dialing: 1. Press **FEATURE *82**. 2. Press # to select the mode. 3. Press **HOLD** to store the mode. **Standard Dial:** Select a line, then dial the number. (Standard Dial is always available, even when another dialing mode is selected.) **Automatic Dial:** Dial the number without choosing a line button first. Your prime line is automatically selected for the call. **Pre-Dial:** Dial the number, then press a line button to place the call. Edit the number by pressing the volume bar before placing the call.

Table 5 Telephone features (Sheet 3 of 7)

Feature	Description
Do Not Disturb	**FEATURE 85** **Cancel: FEATURE #85** When you are not on a call, prevent all incoming calls, except priority calls, from ringing at your telephone. When you are on a call, block an incoming priority call.
Group Listening	**FEATURE 802** **Cancel: FEATURE #802** Use both the handset and speaker while you are on a call. To avoid electronic feedback, keep the handset away from the speaker during the call, and press **RLS** to hang up. **Note:** Most of the portable handsets do not have speakers, and cannot use this feature.
Handsfree	**Handsfree/mute** or **Handsfree** button Press the key to transfer a call from the handset/headset to the telephone speaker. If you lift the handset, return it to the cradle. **Note:** Handsfree speaker volume returns to the default volume set at the telephone at the end of each call.
Hold	Press **HOLD** Temporarily suspend a call. To retrieve a held call, press the line button for the held call. (Press **HOLD** on model 7000, 7100, and 2001 telephones to toggle between two calls.)
Hold - Exclusive	**FEATURE 79** or **FEATURE/HOLD** Temporarily suspend a call and prevent other telephones from picking it up.
Hold - Auto	**FEATURE 73** **Cancel: FEATURE #73** Set your telephone to automatically put a call on Hold when you answer a second call, or stop your telephone from doing so. Default is selected (feature is on). **Note:** Telephones which have system-wide call appearance buttons (SWCA) must have this feature active (selected).
Language choice	**FEATURE *501**: Select Primary Language for the telephone display. **FEATURE *502**: Select Alternate Language for the telephone display. **FEATURE *503**: Select Alternate Language 2 for the telephone display. **FEATURE *504**: Select Alternate Language 3 for the telephone display.
Last Number Redial	**FEATURE 5** Automatically redial the last external telephone number that you dialed.
Line pools	**FEATURE 64** With a line pool, telephones can share several lines for placing calls. 1. Press **FEATURE 64** or an intercom button. 2. Enter a line pool access code. (See your system administrator for a list.)
Line redirection	**FEATURE 84** **Cancel: FEATURE #84** Send calls arriving on an external line to another telephone outside your system. (Some external lines do not support this feature. See your system administrator.) This feature is not available on model 7000, 7100, or 2001 telephones.
Link	**FEATURE 71** Generate a Link signal to access a PBX or other host exchange.

Table 5 Telephone features (Sheet 4 of 7)

Feature	Description
Long tones	**FEATURE 808** Generate a tone for as long as you hold down a button. This is used to communicate with devices such as fax or answering machines. Long tones are in effect only for your current call.
Messages	**FEATURE 1** **Cancel: FEATURE #1** Send a message to another telephone within your system. **To view and reply to your messages:** 1. Press **FEATURE 65**. 2. Press * and # to view your message list. 3. Press 0 to call the person who left you the message. **To erase a message:** Press **HOLD** while viewing a message.
Moving line buttons	**FEATURE *81** Change the position of your line or hunt group buttons. 1. Press **FEATURE *81**. 2. Press the line button that you want to move. 3. Press the button to which you want to move the line. 4. Press **RLS**. The two buttons are exchanged. 5. Update the button label strip on your telephone. Line buttons cannot be exchanged with intercom, answer DN or handsfree buttons.
Mute	**Handsfree/mute** or **Mute** button Press this button when you do not want the caller to hear anything from your side of a handsfree call. The display light beside the button blinks when the call is muted. The mute button on the T-series and i-series telephones mutes all types of calls. **Page announcement note:** A call retrieved from hold after a page announcement does not necessarily remain muted.
Name and number block	**FEATURE 819** Block either the outgoing name, or number, or both for a specific call.
Page	**FEATURE 60** and code (1 to 3) and zone (0 to 6) Make a page announcement through either the internal (code 1) or external (code 2) speakers, or both (code 3). Zone 0 pages all zones. Page announcements are programmed to timeout after a pre-selected amount of time.
	Internal page **FEATURE 61** and zone (0 to 6) Make a page announcement to all, or to a specific group of telephones, through the telephone speakers. Zone 0 pages all zones.
	External page **FEATURE 62** Make a page announcement through an external loudspeaker system.

Table 5 Telephone features (Sheet 5 of 7)

Feature	Description
	Internal and external page **FEATURE 63** and zone (0 to 6) Make a page announcement through both your telephone speakers and an external loudspeaker system. Zone 0 pages all zones.
	Incoming page during active call: The system can be set to either: • Put an active call on hold, and broadcast the incoming page. • Archive the page until you release the call. This feature is set by your system administrator. **Note: Business Series Terminals:** A call on mute when the page comes in, does not remain muted when it is released from hold after the page.
Pause	**FEATURE 78** Program in an external autodial sequence to insert a 1.5-second delay. For pulse dialing: * also inserts a 1.5-second delay. **Note:** This feature is not supported on ISDN trunks.
Priority call	**FEATURE 69** Interrupt a person who is on a call. A person on another call can press **FEATURE 85** (Do Not Disturb) to block priority calls.
Privacy	**FEATURE 83** Change the privacy setting for an external line. If a line normally has privacy, this permits another telephone that shares the line to join your call by selecting the line while you are using it. If a line normally has privacy disabled, this prevents another telephone that shares the line from joining your call by selecting the line while you are using it. The privacy setting is re-established once you end your call or when you enter the Privacy feature code again.
Ring again	**FEATURE 2** **Cancel: FEATURE #2** Monitor a busy or unanswered telephone, or a busy line pool within your system. Ring Again signals you to call back when the telephone or line pool becomes available.
Ring type	**FEATURE *6** Select a distinctive ring to help differentiate between your telephone and others nearby. 1. Press **FEATURE *6**. 2. Enter the ring type number (1 to 4). 3. Press **HOLD**.
Ring volume	**FEATURE *80** Make your telephone ring so that you can adjust the volume. You also can adjust the volume any time your telephone rings.
Run/stop	**FEATURE *9** Store more than one autodial number or external carrier feature code on one memory button by inserting a break point between numbers or codes. The first press of the button dials the first number or code; the next press dials the next number or code. You can program up to four numbers or codes separated by break points.
Saved number redial	**FEATURE 67** Save a number to redial later. Enter the code while you are on a call that you have dialed to save the number. Enter the code when you are not on a call to redial the saved number.

Table 5 Telephone features (Sheet 6 of 7)

Feature	Description
Service schedules	**FEATURE 870** Display the modes that have been turned on at a designated control set.
Ringing services	**FEATURE 871** **Cancel: FEATURE #871** Turn on one of six schedules for alternative ringing/call answering arrangements from a designated control telephone.
Restriction services	**FEATURE 872** **Cancel: FEATURE #872** Turn on one of six services for restrictions on particular lines or telephones from a designated control telephone. You are required to enter a password.
Routing services	**FEATURE 873** **Cancel: FEATURE #873** Turn on one of six services for routing on particular lines or telephones from a designated control telephone. You must enter a password.
Speed dial - using	**FEATURE 0** Dial an external telephone number using a two or three-digit code. There are two types of speed dial codes: system (01-70 or 001 to 255) and personal (71 to 94). System speed dial codes can be used from any display telephone in the system. They are assigned by your system administrator. Personal speed dial codes are used exclusively at your telephone. **To make a call using a speed dial code:** 1. Press **FEATURE 0**. 2. Enter the two- or three-digit code for the number.
Speed dial - programming	**To program personal speed dial numbers:** 1. Press **FEATURE** ***4**. 2. Enter a two-digit code from 71 to 94. 3. Specify the external line by pressing a line button, a line pool button, or the intercom button. If you do not specify the external line, the system automatically chooses a line for the call. 4. Dial the telephone number you want to program (up to 24 digits). 5. Press **HOLD**. 6. Record the code and number you have just programmed. **Note:** You cannot program personal speed dial numbers while someone else is programming your system.
Static time and date	**FEATURE 806** **Cancel: FEATURE #806** Change the first line of the display to the current time and date.
SWCA keys	**FEATURE *521** to **FEATURE *536** programmed to buttons with indicators If you are part of a call group, you may have a number of line buttons that are labelled as SWCA. How you use these buttons depends on how the System Administrator set up the system. (Refer to the SWCA user card for detailed instructions.) **FEATURE *520** Find first available SWCA key assigned to this telephone. **FEATURE *537** Find the oldest parked SWCA call on this telephone. **FEATURE *538** Find the newest parked SWCA call on this telephone.
Time	**FEATURE 803** Briefly display the time and date while you are on a call.

Table 5 Telephone features (Sheet 7 of 7)

Feature	Description
Transfer	**FEATURE 70** Send a call to another telephone within your system, or to an external telephone. You may not be able to transfer a call on an external line to an external telephone, depending on the capabilities of the lines. **Make or answer a call.** 1. Press **FEATURE 70**. 2. Call the person to whom you want to transfer the call. 3. Stay on the line if you wish to speak to the person first. 4. Press **RLS** to complete the transfer. If an external call is transferred to a busy internal or network extension, or is not answered after a few rings, the call automatically rings you back.
Trunk answer	**FEATURE 800** Answer an external call that is ringing on a line that has been placed into a Ringing Service schedule, from any telephone in your system. This feature does not work for a private line.
Voice call	**FEATURE 66** Make a voice announcement or begin a conversation through the speaker of another telephone without first making the other telephone ring.
Voice call deny	**FEATURE 88** **Cancel: FEATURE #88** Prevent your telephone from receiving voice calls. Do Not Disturb (**FEATURE 85**) also prevents your telephone from receiving voice calls.
Wait for dial tone	**FEATURE 804** Program in an external autodial number to cause the system to wait to receive dial tone from another system before proceeding with the dialing sequence.

Call Display Services

The following features are available only if you subscribe to Call Display services from your local telephone company.

Table 6 Call Display Services (Sheet 1 of 2)

Service	Description
Autobumping	**FEATURE 815** **Cancel: FEATURE #815** Enable the system to delete automatically the oldest log item from a full Call Log, so that a new log item can be stored.
Call information	**FEATURE 811** Display the name, number, or line name of a ringing or held call. Press # to move through the information displays.

Table 6 Call Display Services (Sheet 2 of 2)

Service	Description
Call log - view	**FEATURE 812** Call Log displays use the following special characters: • underline: identifies a new item • handset icon: identifies answered calls • globe icon: identifies long distance calls • forward slash: identifies that the information has been shortened **To view your Call Log:** 1. Press **FEATURE 812**. 2. Press * to view old items. Press # to view new items. Press 0 to return to the last viewed item. 3. Press # and * to move through your items. 4. Press the volume bar to view more information on an item.
Call log - erase entry	**To erase a Call Log entry:** 1. Press **HOLD** while viewing an item.
Call log - return call	**To return a call from your Call Log:** 1. Display the desired number on your telephone. 2. Edit the number, if required. You can add numbers for long distance dialing or line pool access or remove numbers using the volume bar. 3. Press a line button. 4. Lift the handset.
Call log - options	**FEATURE *84** Select the type of calls that automatically are stored in your Call Log. Press # to see the next setting. Press **HOLD** to select the displayed setting.
Call log - password	**FEATURE *85** Program a four-digit password for your Call Log. To reset a forgotten password, see your system administrator.
Logit	**FEATURE 813** Store caller information for your current call in your Call Log.

ETSI feature

Table 7 ETSI feature

Feature	Description
MCID (ETSI feature)	**FEATURE 897** must be entered 30 seconds after the caller hangs up, and before you hang up. Record caller information for the last external call at the central office that assigned the line. This feature works only if the incoming calls enter over ETSI ISDN lines, and the feature is activated in programming. Check with your system administrator. ETSI is the European standard. The North American equivalent is ANSI.

Chapter 8
IP telephone overview

IP telephony provides the flexibility, affordability, and expandability of the Internet to the world of voice communications.

This section includes an overview of the components that make up the BCM IP telephony and Voice over IP (VoIP) features:

- "IP telephones and VoIP trunks" on page 64
- "Creating the IP telephony network" on page 65
- "Key IP telephony concepts" on page 68

BCM with VoIP provides several critical advantages:

- **Cost Savings.** IP networks can be significantly less expensive to operate and maintain than traditional networks. The simplified network infrastructure of an Internet Telephony solution cuts costs by connecting IP telephones over your LAN and eliminates the need for dual cabling. Internet Telephony can also eliminate toll charges on site-to-site calls by using your existing IP network. By using the extra bandwidth on your IP network for IP Telephony, you leverage the untapped capabilities of your data infrastructure to maximize the return on your current network investment.
- **Cost flexibility.** The three models of IP telephones offer three levels of functionality, that allow you to choose an IP telephone that fits your budget and/or your service requirements.
- **Portability and flexibility**. Employees can be more productive because they are no longer confined by geographic location. IP telephones work anywhere on the network, even over a remote connection. With Nortel wireless e-mobility solutions, your phone, laptop, or scanner can work anywhere on the network where a an 802.11b access point is installed. Network deployments and reconfigurations are simplified, and service can be extended to remote sites and home offices over cost-effective IP links. As well, IP telephone functionality can be transferred between IP telephones using the Hot desking feature. All your telephone features and setup can travel with you between offices.
- **Simplicity and consistency**. A common approach to service deployment allows further cost-savings from the use of common management tools, resource directories, flow-through provisioning, and a consistent approach to network security. As well, customers can centrally manage a host of multimedia services and business-building applications via a Web-based browser. The ability to network existing PBXs using IP can bring new benefits to your business. For example, the ability to consolidate voice mail onto a single system, or to fewer systems, makes it easier for voice mail users to network.

- **Compatibility**. Internet telephony is supported over a wide variety of transport technologies. A user can gain access to just about any business system through an analog line, Digital Subscriber Line (DSL), a LAN, frame relay, asynchronous transfer mode, SONET, or wireless connection.
- **Scalability**. A future-proof, flexible, and safe solution, combined with high reliability, allows your company to focus on customer needs, not network problems. Nortel internet telephony solutions offer hybrid environments that leverage existing investments in Meridian and Norstar systems.
- **Increased customer satisfaction**. Breakthrough e-business applications help deliver the top-flight customer service that leads to success. By providing your customers with rapid access to sales and support personnel via telephone, the Web, and e-mail, your business can provide better customer service than ever before.

IP telephones and VoIP trunks

This section describes two similar applications for IP telephony on the BCM system: IP telephones and VoIP trunks. These applications can be used separately or together as a network voice/data solution.

Refer to the information under the following headings:

- IP telephones
- "VoIP trunks" on page 65

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 through the integrated interface (LAN card) on the BCM.

Calls made from IP telephones through the BCM can pass over VoIP trunks or across Public Switched Telephone Network (PSTN) lines.

Nortel provides two types of IP telephones. The IP telephones are wired to the IP network using Ethernet, in the case of the i-series IP telephones, 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. H.323 trunks support private networking between BCMs. H.323 trunks can support connections to a number of different types of equipment, including the Meridian 1 (running IPT), Succession 1000/M, DMS100 switches, and SL100 switches, and trunk applications.

Creating the IP telephony network

The following explains the components of the BCM system and the devices it interoperates to create a network.

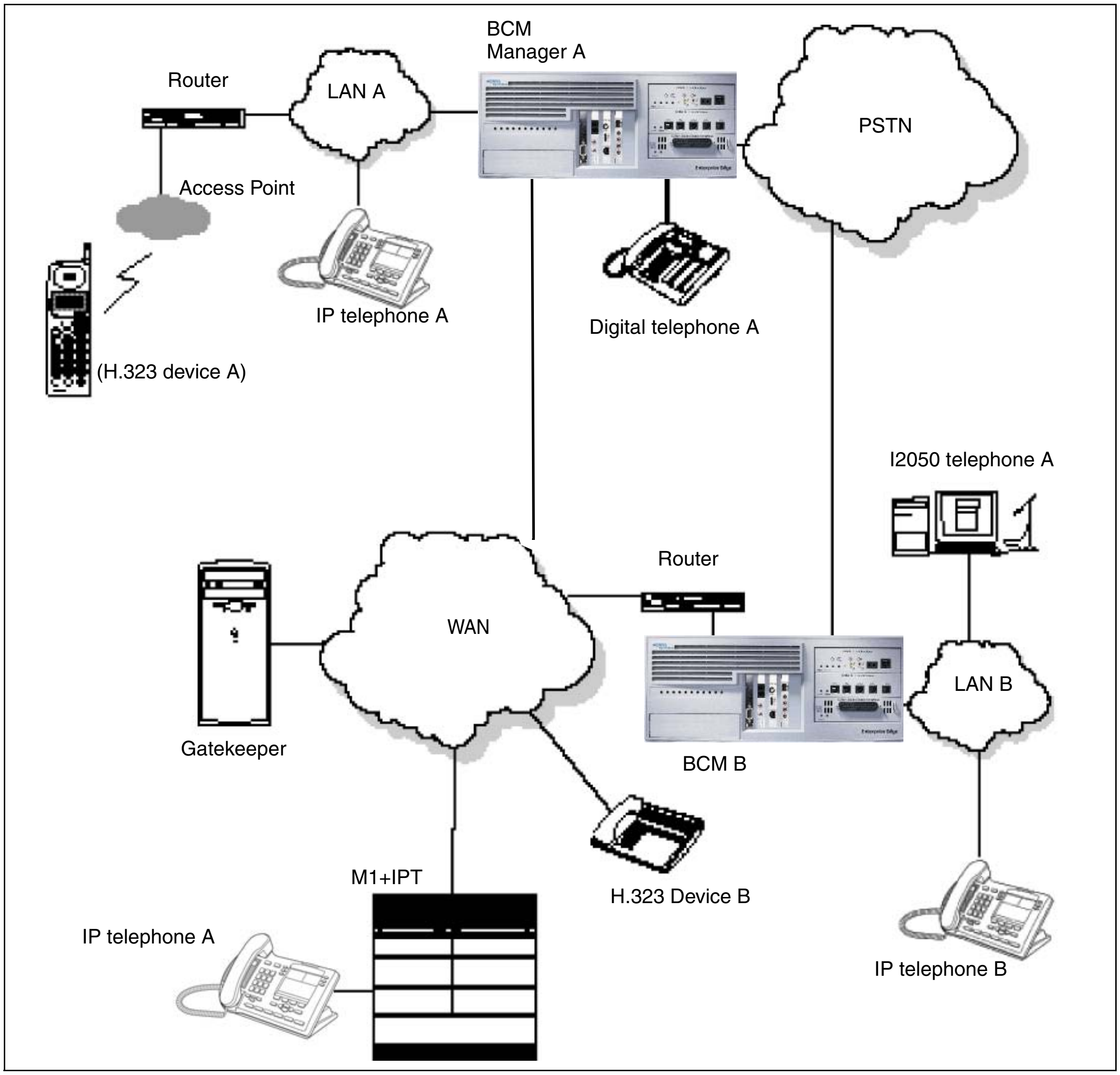
The information under the following headings describes the various components of the system:

- "M1-IPT" on page 67
- "Telephones" on page 67
- "Gatekeepers on the network" on page 67
- "IP network" on page 68
- "Public Switched Telephone Network" on page 68

Figure 11 shows components of a BCM network configuration.

In this example, two BCM systems are connected both through a PSTN connection and through an IP network connection. The IP network connection uses VoIP trunks. If the PSTN connections use dedicated ISDN lines, the two systems have backup private networks to each other. Both BCM systems use VoIP trunks through a common IP network to connect to the Meridian (M1-IPT) system.

Figure 11 Network diagram



Networking with BCM

The BCM is a key building block in creating your communications network. It interoperates with many devices, including the Meridian 1 system and H.323 devices. The BCM system can be connected 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.

The BCM can be connected to a LAN through a the integrated interface LAN card, and to a PSTN through trunk media bay modules, as shown for BCM A in Figure 11. Through these networks, the system accesses other systems and network equipment connected to the network.

M1-IPT

The Meridian 1 Internet Telephony Path (M1-IPT) allows Meridian 1 systems to communicate with the BCM via H.323 trunks. Telephones on the M1, such as Meridian telephone A, can initiate and receive calls with the other telephones on the system across IP networks.

To provide fallback at times when IP traffic cannot pass, you can also connect the Meridian to the BCMs through ISDN PRI SL-1 lines, which provide the same MCDN capability that you can achieve through the H.323 VoIP trunks with MCDN active.

A BCM connected to an M1-IPT using the MCDN protocol can provide access to a central voice mail and call attendant systems, which can streamline multi-office telephony administration.

Telephones

The BCM can communicate using digital telephones (Model 7000, 7100, 7208, 7316, 7316E/7316E+KIMs, 7406 (cordless telephone), Norstar M-series telephones, ISDN telephones, analog telephones, and IP telephones and applications. With this much flexibility, the BCM can provide the type of service you require to be most productive in your business.

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 integrated interface (LAN card) or the IP network and are translated within the BCM to voice channels.

The IP telephones connect to the BCM across an IP network through either a LAN or a WAN. From the BCM connection, they can then use standard lines or VoIP trunks to communicate to other telephones on other public or private networks. The BCM also supports H.323 (version 4) and H.323 third-party devices through this type of connection.

Gatekeepers on the network

A gatekeeper tracks IP addresses of specified devices, and provides routing and (optionally) authorization for making and accepting calls for these 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. Referring to Figure 11, for example: Digital telephone A wants to call IP telephone B, which is attached to BCM B, over a network that is under the control of a gatekeeper. Digital telephone A sends a request to the gatekeeper. The gatekeeper, depending on how it is programmed, provides Digital telephone A with the information it needs to contact BCM B over the network. BCM B then passes the call to IP telephone B.

The BCM does not contain a gatekeeper application. If you want to put a gatekeeper on your network, it must be put on a separate gatekeeper server. The BCM is compatible with CS1000 (CSE1K) gatekeepers.

Warning: Meridian 1 IPT does not support the RadVision gatekeeper.

IP network

In the network shown in Figure 11, several LANs and a WAN are shown. When planning your network, be sure to consider all requirements for a data network. Your network administrator should be able to advise you about the network setup and how the BCM fits into the 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.

If you want to deploy IP telephones that will be connected to a LAN outside of the LAN that the BCM is installed on, you must ensure the BCM is able to communicate across the WAN interface at that location.

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 the integrated interface (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.

Public Switched Telephone Network

The Public Switched Telephone Network (PSTN) can play an important role in IP telephony communications. In many installations, the PSTN forms a fallback route. If a call across a VoIP trunk does not have adequate voice quality, the call can be routed across PSTN lines instead, either on public lines or on a dedicated ISDN connection between the two systems (private network). The BCM also serves as a gateway to the PSTN for all voice traffic on the system.

Key IP telephony concepts

In traditional telephony, the voice path between two telephones is circuit switched. This means that the analog or digital connection between the two telephones is dedicated to the call. The voice quality is usually excellent, since there is no other signal to interfere.

In IP telephony, each IP telephone encodes the speech at the handset microphone into small data packets called frames. The system sends the frames across the IP network to the other telephone, where the frames are decoded and played at the handset receiver. If some of the frames get lost while in transit, or are delayed too long, the receiving telephone experiences poor voice quality. On a properly-configured network, voice quality should be consistent for all IP calls.

The information under the following headings describes some of the components that determine voice quality for IP telephones and trunks:

- "Codecs" on page 69
- "Jitter buffer" on page 69

- "QoS routing" on page 70

Codecs

The algorithm used to compress and decompress voice is embedded in a software entity called a codec (COde-DECode).

Two popular Codecs are G.711 and G.729. The G.711 Codec samples voice at 64 kilobits per second (kbps) while G.729 samples at a far lower rate of 8 kbps.

Voice quality is better when using a G.711 Codec, but more network bandwidth is used to exchange the voice frames between the telephones.

If you experience poor voice quality, and suspect it is due to heavy network traffic, you can get better voice quality by configuring the IP telephone to use a G.729 Codec.

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.

The BCM supports these codecs:

- G.729
- G.723
- G.729 with VAD (Voice Activity Detection)
- G.723 with VAD
- G.711-uLaw
- G.711-aLaw

Jitter buffer

Voice frames are transmitted at a fixed rate, because the time interval between frames is constant. If the frames arrive at the other end at the same rate, voice quality is perceived as good. In many cases, however, some frames can arrive slightly faster or slower than the other frames. This is called jitter, and degrades the perceived voice quality. To minimize this problem, configure the IP telephone with a jitter buffer for arriving frames.

Note: You can only change the jitter buffer on a configured IP telephone if it is online to the BCM, or if Keep DN Alive is enabled for an offline telephone.

This is how the jitter buffer works:

Assume a jitter buffer setting of five frames.

- The IP telephone firmware places the first five arriving frames in the jitter buffer.
- When frame six arrives, the IP telephone firmware places it in the buffer, and sends frame one to the handset speaker.
- When frame seven arrives, the IP telephone buffers it, and sends frame two to the handset speaker.

The net effect of using a jitter buffer is that the arriving packets are delayed slightly in order to ensure a constant rate of arriving frames at the handset speaker.

This delaying of packets can provide somewhat of a communications challenge, as speech is delayed by the number of frames in the buffer. For one-sided conversations, there are no issues. However, for two-sided conversations, where one party tries to interrupt the other speaking party, it can be annoying. In this second situation, by the time the voice of the interrupter reaches the interruptee, the interruptee has spoken (2*jitter size) frames past the intended point of interruption. In cases where very large jitter sizes are used, some users revert to saying *OVER* when they wish the other party to speak.

Possible jitter buffer settings, and corresponding voice packet latency (delay) for the BCM system IP telephones are:

- None
- Small (G.711/G.729: 0.05 seconds)
- Medium (G.711/G.729: 0.09 seconds)
- Large (G.711/G.729: 0.15 seconds)

QoS routing

To minimize voice jitter over low bandwidth connections, the BCM programming assigns specific DiffServ Marking in the IPv4 header of the data packets sent from IP telephones and from IP trunks.

The DiffServ Code point (DSCP) is contained in the second byte of the IPv4 header. DSCP is used by the router to determine how the packets will be separated for Per Hop Behavior (PHB). The DSCP is contained within the DiffServ field, which was known as the ToS field in older versions. The BCM assigns Expedited Forwarding (EF) PHB for voice media packets. On the BCM, these assignments cannot be adjusted.

Chapter 9
Registering Nortel 20XX and 11XX IP telephones

Nortel IP telephones must register with the system to be able to use the call features and system features.

Note: To use system resources efficiently, deregister DNs of IP sets that are not being used.

Determining the registration process

Registering IP telephones to the system is a two-stage process.

Note: Ensure that you have loaded the appropriate keycodes to activate the Nortel IP telephones on your BCM system.

1 Set up the system programming to receive registration under **Configuration > Telephony Resources > IP & Application Sets**

On the **IP Terminal Global Settings** panel:

a Select the **Enable registration** check box.

b If you want the installers to use a single password to configure and register the telephone, select the **Enable global registration password** check box, and then enter a numeric password in the **Global password** field.

c If you want the system to automatically assign DN records to the telephones, select the **Auto-assign DNs** check box.

Note: To automatically configure IP Phones with DNs assigned:

1) Select the **Enable registration** check box.

2) Select the **Enable global registration password** check box.

3) Leave **Global password** field blank.

4) Select the **Auto-assign DNs** check box.

Once the IP Phones are operational, clear the **Enable registration** check box.

Security Note: Turn **Enable registration** and **Auto-assign DNs** off when the telephones are registered. Nortel cautions that leaving your IP registration open and unprotected by a password can pose a security risk.

2 Configure each telephone (“Configuring telephone settings” on page 72).

How you configure the telephones depends on whether DHCP is active on the system.

— If DHCP (Distributed Host Control Protocol) service on the system is active or the Customer DHCP server has been configured to hand out the specific system network details, the IP telephone automatically attempts to find the server.

After you register the telephone to the system, as described in “Registering the telephone to the system” on page 72, the telephone assumes the parameters it receives from the system, which are described in “Configuring telephone settings” on page 72).

— If DHCP is not configured to provide system information, or if you are not using DHCP on your network, you must configure your telephone parameters before the telephone can register to the system. In this case, follow the directions in “Configuring telephone settings” on page 72, and then follow any of the prompts that appear, as described in “Registering the telephone to the system” on page 72.

— If an external DHCP server is not present, the DHCP server on the main unit supplies IP configuration information for all IP devices (PCs and IP Phones). It also supplies specific connection information to the IP Phones.

Registering the telephone to the system

When you first connect the telephone to the IP connection, you receive one of the following:

- If the telephone is not yet registered, and when a password is entered in the Terminal Registration screen, the telephone prompts you for that password.
- If **Auto Assign DN** is not selected, the telephone prompts you for a DN. Refer to “Configuring telephones: IP telephones” in the *BCM 4.0 Device Configuration Guide* (N0060600).
- If you are prompted for a password, enter the password and press OK.
- If you are prompted for a DN, enter the DN you want assigned to this telephone and press **OK**.

When the telephone registers, it downloads the information from the system IP Telephony record to the telephone configuration record. This can include a new firmware download, which occurs automatically. If new firmware downloads, the telephone display indicates the event.

Note: If the telephone displays a prompt that indicates it cannot find the server, follow the instructions in “Configuring telephone settings” on page 72 to enter the specific network path. “Troubleshooting IP telephones” on page 76 describes other possible prompt messages.

Configuring telephone settings

If you are not automatically registered to the system, you can configure the telephone settings to enable you to access a system on the network. You also must perform these steps if your IP telephone is not connected to the same LAN to which the system is connected.

To access the local configuration menu on an IP telephone

2001/2002/2004 phones

1 Restart the telephone by disconnecting the power, then reconnecting the power.
After about four seconds, the top light flashes and `NORTEL NETWORKS` appears on the screen.

2 When the greeting appears, **immediately, and quickly** press the four display buttons one at a time, from left to right. See Figure 12. These buttons are located directly under the display.

2033 phone

1 Restart the telephone by disconnecting the power, then reconnecting the power.
After about four seconds, the top light flashes and `NORTEL NETWORKS` appears on the screen.

2 When the greeting appears, **immediately, and quickly** press the three display buttons one at a time, from left to right. See Figure 12. These buttons are located directly under the display.

2007 phone

1 Restart the telephone by disconnecting the power, then reconnecting the power.
After about four seconds, `NORTEL` appears on the screen.

2 When the greeting appears, **immediately, and quickly** press 007* on the dialpad.

Using the dialpad

1 Tap the tool icon. See Figure 12.

2 When prompted for a password, using the dialpad enter COLOR*SET (26567*738).

1120E/1140E phones

1 Restart the telephone by disconnecting the power, then reconnecting the power.
After about four seconds, the top light flashes and `NORTEL NETWORKS` appears on the screen.

2 When the greeting appears, **immediately, and quickly** press the four display buttons one at a time, from left to right. See Figure 12. These buttons are located directly under the display.

Using the dialpad

1 Press the **Services** key twice, quickly. See Figure 12.

2 Use the navigation keys to find the service to modify.

3 Press **Select.**

Figure 12 IP Phones

Press the button sequence within 1.5 seconds; otherwise the telephone does not enter configuration mode.

- If `Manual Cfg DHCP(0 no, 1 yes)` appears on the screen, you successfully accessed the configuration mode.
- If any other message appears, disconnect, then reconnect the power, and try to access the configuration mode again.

4 Enter the network parameters, as prompted.

As each parameter prompt appears, use the keypad to define values.

Use the * key to enter the period in the IP addresses.

Press **OK** to move forward.

Table 8 describes the values for each display parameter.

Table 8 IP telephone server configurations (Sheet 1 of 3)

Field	Value	Description
DHCP	0 or 1	Enter 0 if your network is not using a DHCP server to dispense IP addresses. (Partial DHCP) Enter 1 if your network does use a DHCP server. If you choose to use a DHCP server rather than allocating static IP addresses for the IP telephones, skip the remainder of this section.

Table 8 IP telephone server configurations (Sheet 2 of 3)

Field	Value	Description
If DHCP = 0		
SET IP	<IP address>	The set IP must be a valid and unused IP address on the network to which the telephone is connected.
NETMASK	<subnet mask address>	This is the subnet mask. This setting is critical for locating the system to which you want to connect.
DEF GW	<IP address>	Default Gateway on the network (for example, the nearest router to the telephone. The router for IP address W.X.Y.Z is usually at W.X.Y.1). If there are no routers between the telephone and the system network adaptor to which it is connected, (for example, a direct HUB connection), then enter the Published IP address of the BCM as the DEF GW. If the IP telephone is not connected directly to the Published IP address network adapter, set the DEF GW to the IP address of the network adaptor to which the telephone is connected.
Emulation Key Mapping	0 or 1	0 = Handset 1= Handsfree Default setting is 1 (handsfree) and should not be changed. **Note**: This setting applies to the 2033 model only.
If DHCP = 1		
Manual Cfg? DHCP:	Full = 0 Partial = 1	If you indicate DHCP for the telephone, but you want to enter static IP addresses, choose 1 (Partial). If you choose 0 (Full), the DHCP server assigns IP addresses that are not static.
If DHCP = 0 or Partial		
S1 IP	<IP address>	This is the Published IP address of the first system to which you want to register the telephone.
S1 PORT	Default: *7000	This is the port the telephone uses to access this system.
S1 ACTION	Default: 1	
S1 RETRY COUNT	<digits between 0 and 255>	Set this to the number of times you want the telephone to retry the connection to the system.
S2 IP	<IP address>	This is the Published IP address of the second system to which you want to register the telephone. It can be the same as the S1 setting.
S2 PORT	Default: *7000	This is the port the telephone uses to access this system.
S2 ACTION	Default: 1	
S2 RETRY COUNT	<digits between 0 and 255>	Set this to the number of times you want the telephone to retry the connection to the system.
VLAN	0: No VLAN 1: Manual VLAN 2: Automatically discover VLAN using DHCP	Choose 0: NO VLAN if there is no VLAN on the network. If you do not have DHCP on the network, or if DHCP is supplied by a remote server, select number 1 and enter the VLAN ID*. If you have the system DHCP active on your system, select number 2 if you want DHCP to find the VLAN assignment automatically. *VLAN is a network routing feature provided by specific types of switches. To find out if VLAN has been deployed on your system, check with your network administrator. If VLAN is deployed, the system administrator responsible for the switch can provide the VLAN IDs for your system.

Table 8 IP telephone server configurations (Sheet 3 of 3)

Field	Value	Description
Cfg XAS?	0: No (default) 1: Yes	If you want to enable connection to a Net6 service provider server, choose 1. You are then prompted for an IP address for the server.
* **Firewall note:** Ensure that the firewall filters are set up to allow IP traffic into and out of the system.		

After you have entered all the configuration information, the telephone attempts to connect to the system. The message `Locating Server` appears on the display. If the connection is successful, the message changes to `Connecting to Server` after about 15 seconds. Initialization can take several minutes. Do not disturb the telephone during this time.

When the telephone connects to the server and is ready to use, the display shows the time and date. As well, the six keys at the top of the display are labelled.

If you experience problems with IP telephone registration, refer to the section: "Troubleshooting IP telephones" on page 76.

Notes:

- If the DN record is not configured yet, as is the case with auto-assigned DNs, you can only place local calls until other lines are assigned in the DN record.
- If the telephone has not been registered before, you receive a `New Set` message. Enter the information, as prompted. Refer to "Registering the telephone to the system" on page 72.

Troubleshooting IP telephones

If the system is not properly configured, several messages can appear.

Table 9 IP telephony display messages

Message	Description/Solution
`SERVER: NO PORTS LEFT`	The system has run out of ports. This message remains on the display until a port becomes available and the telephone is powered down and then up. To obtain more ports, you can install additional VoIP keycodes.
`Invalid Server Address`	The S1 is incorrectly configured with the IP address of a system network adapter other than the published IP address.
`IP Address conflict`	The telephone detected that a device on the network is currently using the IP address allocated to the telephone.
`Registration Disabled`	The Registration on the system is set to OFF.
`SERVER UNREACHABLE.` `RESTARTING . . .`	Check that you have entered the correct Netmask and gateway IP addresses. If the settings are correct, contact your system administrator.
`NEW SET`	The telephone has not been connected to the system before, and must be registered.

Programming note: To display the configuration information for a telephone connected to the system:

- If the telephone is engaged, press the key, followed by the key.

Operation issues

Table 10 provides solutions to potential problems.

Table 10 IP telephone troubleshooting

Problem	Suggested solution or cause
Telephone does not connect to system	If an IP telephone does not display the text `Connecting to server` within two minutes after power up, the telephone did not establish communications with the system. Double-check the IP configuration of the telephone and the IP connectivity to the system (cables, hubs, and so on).
Slow connection between the handset and the system	If the connection between the IP client and the system is slow (ISDN, dialup modem), change the preferred CODEC for the telephone from G.711 to G.729. See Table 8.
One-way or no speech paths	Signaling between the IP telephones and the system uses the system port 7000. However, voice packets are exchanged using the default RTP ports 28000 through 28255 at the BCM, and ports 51000 through 51200 at the IP telephones. If these ports are blocked by the firewall or NAT, you will experience one-way or no-way speech paths.
Change the contrast level	When an IP telephone is connected for the first time, the contrast level is set to the default setting of 1. Use **FEATURE *7** and the `UP` or `DOWN` key to adjust the contrast.
Block individual IP sets from dialing outside the system.	If you want to block one or more IP telephones from calling outside the system, use Restriction filters, and assign them to the telephones you want to block. Restriction filters are set up under **Configuration > Telephony > Call Security > Restriction Filters**.

Deregistering IP telephones

You can deregister selected IP telephones from the system, and force the telephone to go through the registration process again.

Note: To use system resources efficiently, deregister DNs of IP sets that are not being used.

Warning: After this feature is activated, all active calls are dropped.

To deregister a IP telephone from the IP record

1 You can access the deregister button from two locations:

- **Configuration > Resources > Telephony Resources > IP & Applications Sets > IP Terminal Details** tab
- **Telephony > Sets > Active Sets > IP Terminal Details tab**

2 Select the IP telephone that you want to deregister.

3 Click **Deregister DN**.

4 Reregister the telephone, as described in "Determining the registration process" on page 71.

Warning: After this feature is activated, all active calls are dropped.

Next step

See IP-specific features: "Global VoIP features" in the *BCM 4.0 Device Configuration Guide* (N0060600).

See Nortel IP telephones user cards.

Chapter 10
Relocating telephones

The following explains how you can physically move a telephone within the system so that the telephone programming follows the telephone to the new location.

- "Moving digital telephones" on page 79
- "Moving IP telephones" on page 80
- "User card list" on page 81 provides a list of the user cards that provide information about using individual types of telephones, and the features they can access.

Moving digital telephones

To move a digital telephone to a new location within the system so that the programmed settings are retained, set relocation (automatic telephone relocation) must be enabled in system programming. Set relocation saves the internal numbers, autodial settings, and personal speed dial codes within the telephone when the telephone is unplugged.

Note: The set relocation feature applies to the digital telephones and analog telephones, only. IP telephones always retain their programming. Refer to "Moving IP telephones" on page 80.

Tips (if set relocation is enabled)
Relocate existing telephones before new telephones are installed on the jacks. This allows the moved telephones to retain their programming.

Plugging a new telephone into a jack from which another telephone was removed, before the original telephone is reconnected to another jack, results in the programming transferring to the new telephone. In this case, when the original telephone is plugged into another jack, it receives default programming, or the programming specifically entered for the DN record that corresponds to the new jack.

When changing a telephone internal number (DN record), wait one minute for automatic telephone relocation to complete its cycle. When you relocate a telephone, the telephone must remain installed and connected in the new location for at least three minutes for the programming relocation to be complete. Moving the telephone again before the three-minute period is up can result in loss of programming.

To enable Set relocation and relocate digital telephones

1. In the Element Manager, click the keys located beside **Configuration > Telephony > Global Settings > Feature Settings**.
2. Select the **Set relocation** check box.

3 **Move the telephone**: Unplug the telephone, and plug it in again at another location. It can take up to 45 seconds for the system to recognize the telephone.

- Clear the **Set relocation** check box after you complete all required moves.

Keeping an IP telephone active

In some circumstances, you may want to have your IP telephone stay active after it is physically disconnected. For example, when your i2050 software phone is turned off, you may still want callers to go to your voicemail. To keep your IP telephone active and retain DN-specific features, activate the **Keep DN alive** feature.

To keep an IP telephone active after it is disconnected

1. In the Element Manager, go to **Configuration > Telephony > Sets > Active Sets**.
2. Click the **Capabilities and Preferences** tab, and IP Terminal details.
3. Select the **Keep DN alive** check box.

> **Note:** Clearing the check box allows the DN record to become inactive if the IP telephone is disconnected.

Moving IP telephones

IP telephones retain their DN when they are moved to a new location on the same subnet. The following instructions apply to Nortel IP telephones.

To move an IP telephone without changing the DN

1. Disconnect the power from the IP telephone or 3-port switch.
2. Disconnect the network connection.
3. At the new location, reconnect the network cable and the power connection.
4. If the new location is on a different subnet, you must make the appropriate changes to the telephone IP addressing. However, do not change the S1 IP address or the S2 IP address. Disconnect the power from the IP telephone or 3-port switch.

> **Note:** If your network is using partial DHCP, reconfiguration is not required at this step.

To move a Nortel IP telephone and change the DN

1. Deregister the DN.
2. Disconnect the network connection and the power connection from the telephone.

3 Reinstall the telephone at the new location, and reconfigure the telephone.

User card list

The following is a list of feature and device user guides that can be found on your system CD:

- *IP Phone 2001 User Guide* (N0027313)
- *IP Phone 2002 User Guide* (N0027300)
- *IP Phone 2004 User Guide* (N0027284)
- *IP Phone 2007 User Guide* (N0064498)
- *IP Phone 1120E User Guide* (NN-10300-062)
- *IP Phone 1140E User Guide* (NN-10300-064)
- *IP Audio Conference Phone 2033 User Guide* (N0060623)
- *i2050 Software Phone Installation Guide* has on-line user help
- *Telephone Feature User Guide*

Appendix A
ASM8, ASM8+, and GASM wiring chart

Analog telephony devices, such as single-line telephones, modems, and fax machines, are connected to the analog station module (ASM) through the RJ-21 connector on the front of the media bay module (MBM) (see Figure 13).

Figure 13   ASM RJ-21 connector

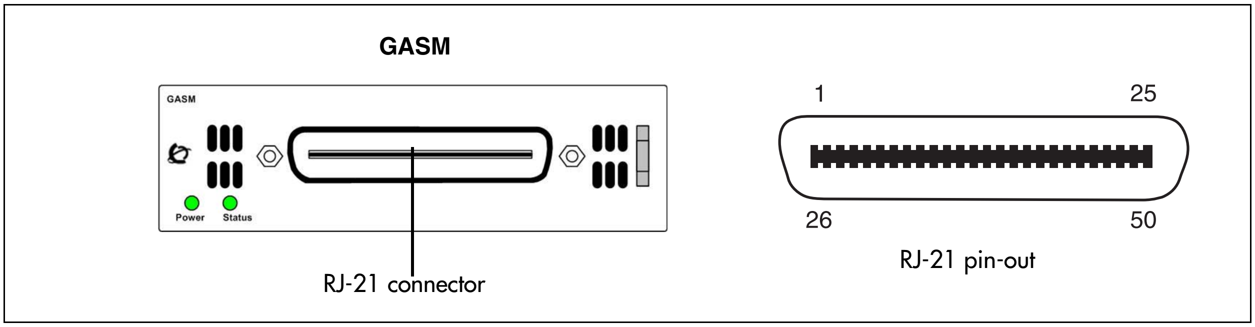


Table 11 lists the wiring details for the RJ-21 connector on the ASM.

Table 11   ASM RJ-21 connector wiring (Sheet 1 of 2)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1	Default DN on Expansion port 2
1	26	Tip	White-Blue	**237**	**269**
	1	Ring	Blue-White		
2	27	Tip	White-Orange	**238**	**270**
	2	Ring	Orange-White		
3	28	Tip	White-Green	**239**	**271**
	3	Ring	Green-White		
4	29	Tip	White-Brown	**240**	**272**
	4	Ring	Brown-White		
5	30	Tip	White-Slate	**241**	**273**
	5	Ring	Slate-White		
6	31	Tip	Red-Blue	**242**	**274**
	6	Ring	Blue-Red		
7	32	Tip	Red-Orange	**243**	**275**
	7	Ring	Orange-Red		
8	33	Tip	Red-Green	**244**	**276**
	8	Ring	Green-Red		
—	34	No connection	Red-Brown	—	—
	9	No connection	Brown-Red		

Table 11 ASM RJ-21 connector wiring (Sheet 2 of 2)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1	Default DN on Expansion port 2
...	...	...	...	...	...
—	50	No connection	Violet-Slate	—	—
	25	No connection	Slate-Violet		

Appendix B
DSM16 and DSM32 wiring charts

Digital telephones, such as the Business Series Telephones, are connected to a digital station module (DSM16 or DSM32) through the RJ-21 connectors on the front of the media bay modules (MBM). The DSM16 has a single RJ-21 connector and the DSM32 has two RJ-21 connectors (see Figure 14).

Figure 14 DSM16 and DSM32 RJ-21 connectors

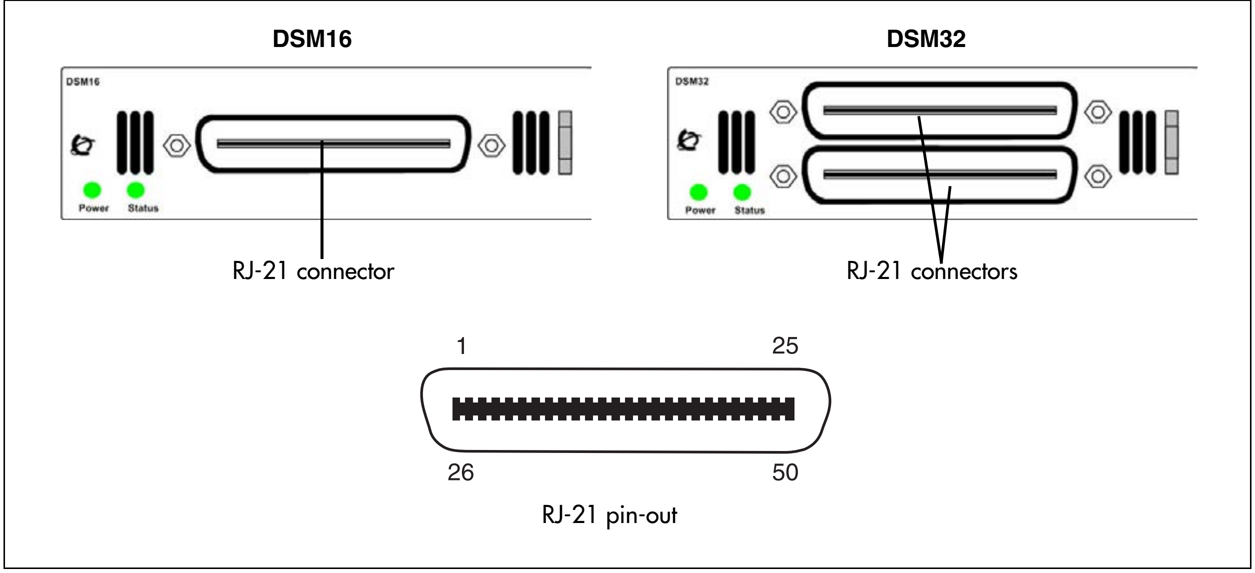


Table 12 lists the wiring details for the RJ-21 connectors on the DSM16 and DSM32.

Table 12 DSM16 and DSM32 RJ-21 connector wiring (Sheet 1 of 3)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1				Default DN on Expansion port 2			
				DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port	DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port
1	26	Tip	White-Blue	237	501	253	601	269	701	285	801
	1	Ring	Blue-White								
2	27	Tip	White-Orange	238	502	254	602	270	702	286	802
	2	Ring	Orange-White								
3	28	Tip	White-Green	239	503	255	603	271	703	287	803
	3	Ring	Green-White								

Table 12 DSM16 and DSM32 RJ-21 connector wiring (Sheet 2 of 3)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1				Default DN on Expansion port 2			
				DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port	DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port
4	29	Tip	White-Brown	240	504	256	604	272	704	288	804
	4	Ring	Brown-White								
5	30	Tip	White-Slate	241	505	257	605	273	705	289	805
	5	Ring	Slate-White								
6	31	Tip	Red-Blue	242	506	258	606	274	706	290	806
	6	Ring	Blue-Red								
7	32	Tip	Red-Orange	243	507	259	607	275	707	291	807
	7	Ring	Orange-Red								
8	33	Tip	Red-Green	244	508	260	608	276	708	292	808
	8	Ring	Green-Red								
9	34	Tip	Red-Brown	245	509	261	609	277	709	293	809
	9	Ring	Brown-Red								
10	35	Tip	Red-Slate	246	510	262	610	278	710	294	810
	10	Ring	Slate-Red								
11	36	Tip	Black-Blue	247	511	263	611	279	711	295	811
	11	Ring	Blue-Black								
12	37	Tip	Black-Orange	248	512	264	612	280	712	296	812
	12	Ring	Orange-Black								
13	38	Tip	Black-Green	249	513	265	613	281	713	297	813
	13	Ring	Green-Black								
14	39	Tip	Black-Brown	250	514	266	614	282	714	298	814
	14	Ring	Brown-Black								
15	40	Tip	Black-Slate	251	515	267	615	283	715	299	815
	15	Ring	Slate-Black								
16	41	Tip	Yellow-Blue	252	516	268	616	284	716	300	816
	16	Ring	Blue-Yellow								
—	42	No connection	Yellow-Orange	—		—		—		—	
	17	No connection	Orange-Yellow								
...	...	...	...	...		...		...		...	

Table 12 DSM16 and DSM32 RJ-21 connector wiring (Sheet 3 of 3)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1				Default DN on Expansion port 2			
				DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port	DSM16 or Lower DSM32 RJ-21	Port	Upper DSM32 RJ-21	Port
—	50	No connection	Violet-Slate	—		—		—		—	
	25	No connection	Slate-Violet								

Appendix C
DTM wiring chart

The digital telephone line is connected to the digital trunk module (DTM) through the RJ-48C jack on the front of the media bay module (MBM) (see Figure 15).

Figure 15 DTM RJ-48C port

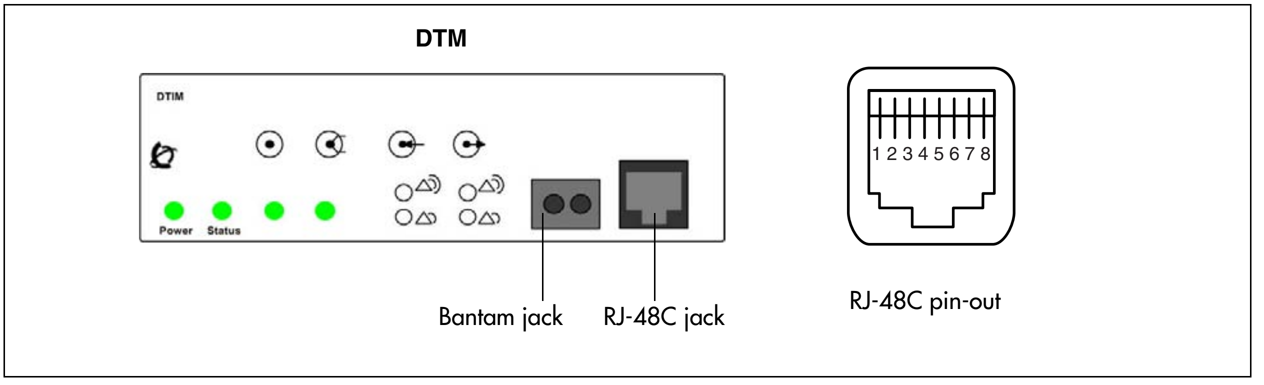


Table 13 and Table 14 list the wiring details for the RJ-48C port.

Table 13 DTM RJ-48C port wiring

Pin	Signal
1	Receive Ring
2	Receive Tip
3	Receive Shield
4	Transmit Ring
5	Transmit Tip
6	Transmit Shield
7	No connection
8	No connection

Table 14 DTM line numbering

Line type	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
T1	065 – 088	095 – 118
PRI	065 – 087	095 – 117
E1	065 – 094	095 – 124

Appendix D
BRIM wiring chart

The digital BRI ISDN lines are connected to the BRIM through the RJ-45 jacks on the front of the media bay module (MBM) (see Figure 16). You can connect up to four BRI ISDN lines to the BRIM.

Figure 16, Table 15, and Table 16 apply to S-Loop and T-Loop connections. S-Loop connections are used to connect S-Loop devices, such as video phones, terminal adapters, and group 3 fax machines. The T-Loop connections are used to connect to the CO/PSTN.

Warning: For a U-Loop connection, the BRIM must be connected only to an NT1 provided by the service provider. The NT1 must provide a Telecommunication Network Voltage (TNV) to Safety Extra Low Voltage (SELV) barrier.

Figure 16 BRIM RJ-45 ports

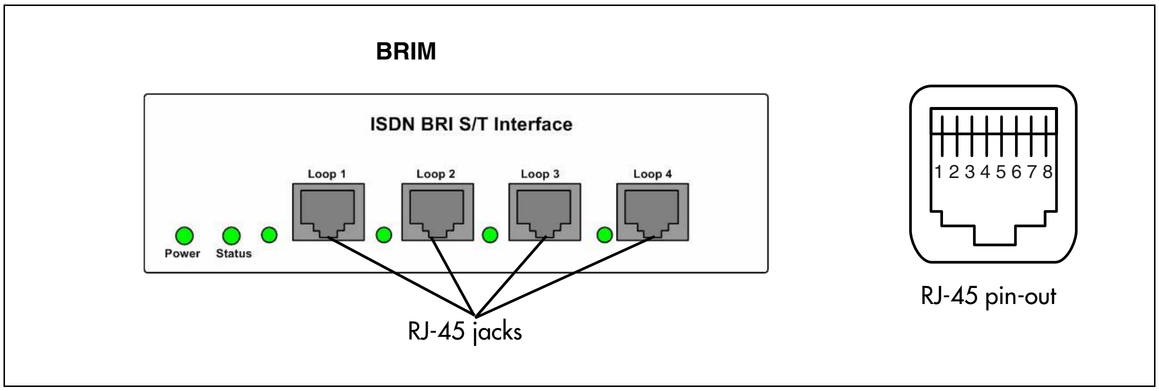


Table 15 and Table 16 list the wiring details for the RJ-45 ports.

Table 15 BRIM RJ-45 port wiring

Pin	Signal	Signal on system side
1	No connection	No connection
2	No connection	No connection
3	+ Receive (+Rx)	+Tx
4	+ Transmit (+Tx)	+Rx
5	- Transmit (-Tx)	-Rx
6	- Receive (-Rx)	-Tx
7	No connection	No connection
8	No connection	No connection

Table 16 BRIM line numbering

Port number	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
1	065 – 066	095 – 096
2	067 – 068	097 – 098
3	069 – 070	099 – 100
4	071 – 072	101 – 102

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