Telephony Device Installation Guide

BCM50 3.0

Business Communications Manager

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Getting started with telephony devices

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About this guide

This guide provides task-based information on how to install analog, digital, IP, and ISDN devices running on a BCM50 system.

Use Element Manager, Startup Profile, and Telset Administration to configure certain BCM50 parameters.

The information in this guide explains

- installing and configuring components
- registering and relocating telephones and devices
- programming loops, configuring digital telephones
- managing system-wide call appearance (SWCA) keys
- setting up central answering positions (CAP)

Audience

This guide is intended for installers responsible for installing, configuring, and maintaining telephony devices on BCM50 systems.

To use this guide, you must

- be an authorized BCM50 installer/administrator within your organization
- know basic Nortel BCM50 terminology
- be knowledgeable about telephony and IP networking technology

About BCM50

The BCM50 system provides private network and telephony management capability to small and medium-sized businesses.

The BCM50 system

- integrates voice and data capabilities, voice over Internet protocol (VoIP) gateway functions, and Quality of Service (QoS) data-routing features into a single telephony system
- enables you to create and provide telephony applications for use in a business environment

BCM50 key hardware elements

BCM50 includes the following key elements:

- BCM50 main unit
- BCM50 expansion unit (compatible with BCM50 main unit)
- media bay modules (MBM):
  - 4 x 16
  - ADID4
  - ADID8
  - ASM8, ASM8+
  - BRIM
  - CTM4, CTM8
  - DSM16, DSM32
  - DSM16+, DSM32+
  - DTM
  - GASM
  - GATM4, GATM8
  - G4 x 16
  - G8 x 16
  - R2MFC

BCM50 features

BCM50 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)

BCM50 applications

BCM50 3.0 supports many applications provided on the existing BCM50 platform.

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 (computer telephony engine) CTE
- VEWAN
- Interactive Voice Response (IVR)
- IP Music
- Contact Center

Symbols and text conventions

These symbols are used to Highlight critical information for the *Administration Guide* 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: A 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 BCM50 main unit and expansion unit power cords from the ac outlet before performing any maintenance procedure.

These conventions and symbols are used to represent the Business Series Terminal display and key pad.

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.

These 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 need to 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 document refers to other related publications, which appear in the following list. To locate specific information, you can refer to the *Master Index of BCM50 3.0 Library*.

Device Configuration Guide (NN40020-300)

Installation and Maintenance Guide (NN40020-302)

Telephone Features User Guide (NN40020-100)

How to get Help

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

Getting Help from the Nortel Web site

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

http://www.nortel.com/support

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

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

Getting Help over the phone from a Nortel Solutions Center

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

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

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

http://www.nortel.com/callus

Getting Help from a specialist by using an Express Routing Code

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

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

Chapter 2
Device description

This chapter describes the telephony devices (telephones) that BCM50 supports.

Analog devices

BCM50 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). To connect a standard analog voice device or data communication device to the BCM50 system through a digital station module, you must install an ATA2 (see Chapter 4, "Installing the analog terminal adapter).

Digital devices

BCM50 supports the following digital devices:

- **T7000**(International only): four memory buttons, without display or indicators
- **T7100**: one-line display, one memory button without indicator
- **T7208**: 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.
- **T7406E cordless handset:** six memory buttons with indicators and a three-line display with three display buttons.

  The T7406E provides cordless mobility in a small office environment. Each base station supports four handsets. 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.
- **Central Answering Position (CAP/eCAP):** provides additional auto dial positions or additional line appearances. The CAP consists of a T7316E telephone and from one to nine key indicator modules (KIMs).

Wireless devices

BCM50 supports the following wireless devices:

- **Dect 413x series handsets:** three display softkeys, four-line handset display, text messaging
- **Dect 414x series handsets:** 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 BCM50 user)

  The handsets communicate with the BCM50 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

BCM50 supports the following IP devices:

- **IP Phone 2001:** connects through an IP link to the BCM50 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 BCM50 line.
- **IP Phone 2002:** connects through an IP link to the BCM50 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 BCM50 line.
- **IP Phone 2004:** connects through an IP link to the BCM50 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 BCM50 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 Key Expansion Module (KEM):** 24 programmable keys (with labels) for IP Phone 2002 or 2004 models; maximum of four IP KEMs for one phone
- **IP Softphone 2050:** provides Voice over IP (VoIP) services using a telephony server and your company's local area network (LAN). One-click direct dialing from various windows and applications. It has twelve user-defined feature keys and four soft keys.
- **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 1110:** graphical, high-resolution LCD display, backlit, with adjustable contrast. It also has three user-defined feature keys and four soft keys.
- **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 brings voice and data to the desktop by connecting directly to a local area network (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 through an Ethernet connection.

  The IP Phone 1140 brings voice and data to the desktop by connecting directly to a LAN ISDN devices
- **Expansion module IP Phone 1100 Series:** Compatible with the IP Phone 1120E and 1140E, this expansion module has 18 self-labelling keys. Up to three modules can be connected to a phone for a maximum of 54 additional line/feature keys.
- **IP Phone 1200 series:** The IP Phones 1210/1220/1230 bring voice and data to the desktop by connecting directly to a local area network (LAN) through an Ethernet connection. Programmable button labels appear beside the keys, and soft key labels appear directly above the keys.
- **Expansion modules for IP Phone 1200 series:** There are two expansion module models for the IP Phone 1200 series of phones: the Expansion Module for IP Phone 1200 Series with display and the Expansion Module for IP Phone 1200 Series with paper label. The expansion modules for IP Phone 1200 Series are hardware accessories that connect to the IP Phone and provide additional line appearances and feature keys. The expansion modules provide either 12 or 18 additional line/programmable feature keys for your IP Phone. Up to seven Expansion Modules for IP Phone 1200 Series with display are supported on an IP Phone, or up to two Expansion Modules for IP Phone 1200 Series with paper labels are supported on an IP Phone. The two expansion module types are not supported on the same IP Phone.

Refer to Chapter 5, "ISDN overview for information on ISDN devices (hardware).

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 3.6	BCM 3.7	BCM 4.0	BCM50 1.0	BCM50 2.0	BCM50 3.0	MBM
T7000 (EU only)	X	X	X	X	X	X	DSM
T7100	X	X	X	X	X	X	DSM
T7208	X	X	X	X	X	X	DSM
T7316	X	X	X	X	X	X	DSM
T7316E	X	X	X	X	X	X	DSM
T7406 (North America only)	X	X	X	X	X	X	DSM
T7406E						X	DSM
T 24 KIM	X	X	X	X	X	X	DSM
M7208	X	X	X	X	X	X	DSM
M7324	X	X	X	X	X	X	DSM
M7406	X	X	X	X	X	X	DSM
Central Answering Position (CAP)	X	X	X	X	X	X	DSM
BST Doorphone	X	X	X		X	X	DSM
Dect 413x		X	X		X	X	DSM
Dect 414x		X	X		X	X	DSM
Digital Mobility Phone 7420					X	X	DSM
Digital Mobility Phone 7430		X			X	X	DSM
Digital Mobility Phone 7440		X			X	X	DSM
IP Phone 1110			X			X	
IP Phone 1120E			X		X	X	
IP Phone 1140E			X		X	X	
Expansion Module IP Phone 1100 Series						X	
IP Phone 1210						X	
IP Phone 1220						X	
IP Phone 1230						X	
IP Phone 2001	X	X	X	X	X	X	
IP Phone 2002	X	X	X	X	X	X	
IP Phone 2004	X	X	X	X	X	X	
IP Phone 2007		X	X		X	X	
IP KEM			X		X	X	
IP Softphone 2050	X	X	X	X	X	X	
IP Audio Conference Phone 2033			X		X	X	
WLAN 2210 Handset		X	X		X	X	

Table 1 Telephony devices release compatibility matrix

Device	BCM 3.6	BCM 3.7	BCM 4.0	BCM50 1.0	BCM50 2.0	BCM50 3.0	MBM
WLAN 2211 Handset		X	X		X	X	
WLAN 2212 Handset			X		X	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
PRI digital lines on a T1 carrier (NA)	DTM	23
PRI digital lines on an E1 carrier (EMEA)	DTM	30
Digital	G4x16	16
Digital	G8x16	16
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)
Analog lines	ADID 4	4
Analog lines	ADID 8	8
Analog lines	G4x16	4
Analog lines	G8x16	8
BRI ISDN lines	BRIM S/T	4 ISDN loops (to a maximum of 8 lines)
R2MFC lines on an E1 carrier	R2MFC	

Table 3 MBM station requirements (Sheet 1 of 2)

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
Digital extensions	G4x16	16

Table 3 MBM station requirements (Sheet 2 of 2)

Type of extension	Type of MBM	Number of extensions per MBM
Digital extensions	G8x16	16
Analog extensions	ASM8	8
Analog extensions	ASM8+	8
Analog extensions	GASM8	8
Analog extensions	G4x16	4
Analog extensions	G8x16	8
Cordless handsets (DECT) (selected profiles only)	DSM32/DSM32+	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

You can connect a a maximum of eight analog telecommunication devices to the analog station media bay modules (ASM8, ASM8+, and GASM). 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] according to EIA/TIA 464)—indicates to the attached device, in an established communication, that the connected device must release the call

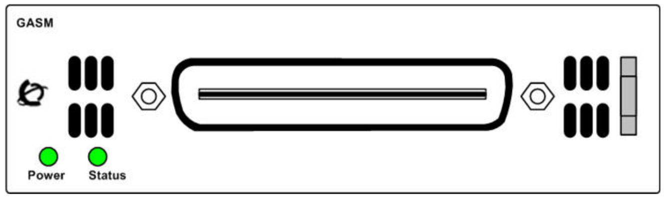
Note: When disconnect happens from the central office (CO), the ASM8+ provides an OSI to the off-hook station of 850 ms (TIA/EIA 464 section 5.4.10.2.4; minimum is 600 ms) as a disconnect signal. If the station remains on-hook after the disconnect signal, the ASM8+ disconnects the station equipment from 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).

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

- 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 two REN/65 V rms per port.
- GASM8—designated as an on-premise station (OPS) port

The ASM8, ASM8+, and GASM each have one RJ-21 connector on the faceplate. Figure 1 on page 20 shows the GASM faceplate.

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 sections for information on installing and configuring an ASM:

- "Installing and configuring a media bay module" on page 20
- "Configuring the media bay module" on page 21
- "Wiring the ASM" on page 22
- "Installing analog devices" on page 22

For more detailed information on installing the BCM50 system and related components, refer to *Installation and Maintenance Guide* (NN40020-302).

Installing and configuring a media bay module

You can install media bay modules (MBM) in BCM50 main units and expansion units, depending on your system requirements.

The primary tasks to install an MBM are

- selecting an MBM for your system
- assigning DS30 resources
- setting MBM dip switches
- installing an MBM

For more detailed information on installing an MBM, refer to *Installation and Maintenance Guide* (NN40020-302).

Configuring the media bay module

For information on installing a media bay module (MBM) and setting the dip switches, refer to the *Installation and Maintenance Guide* (NN40020-302).

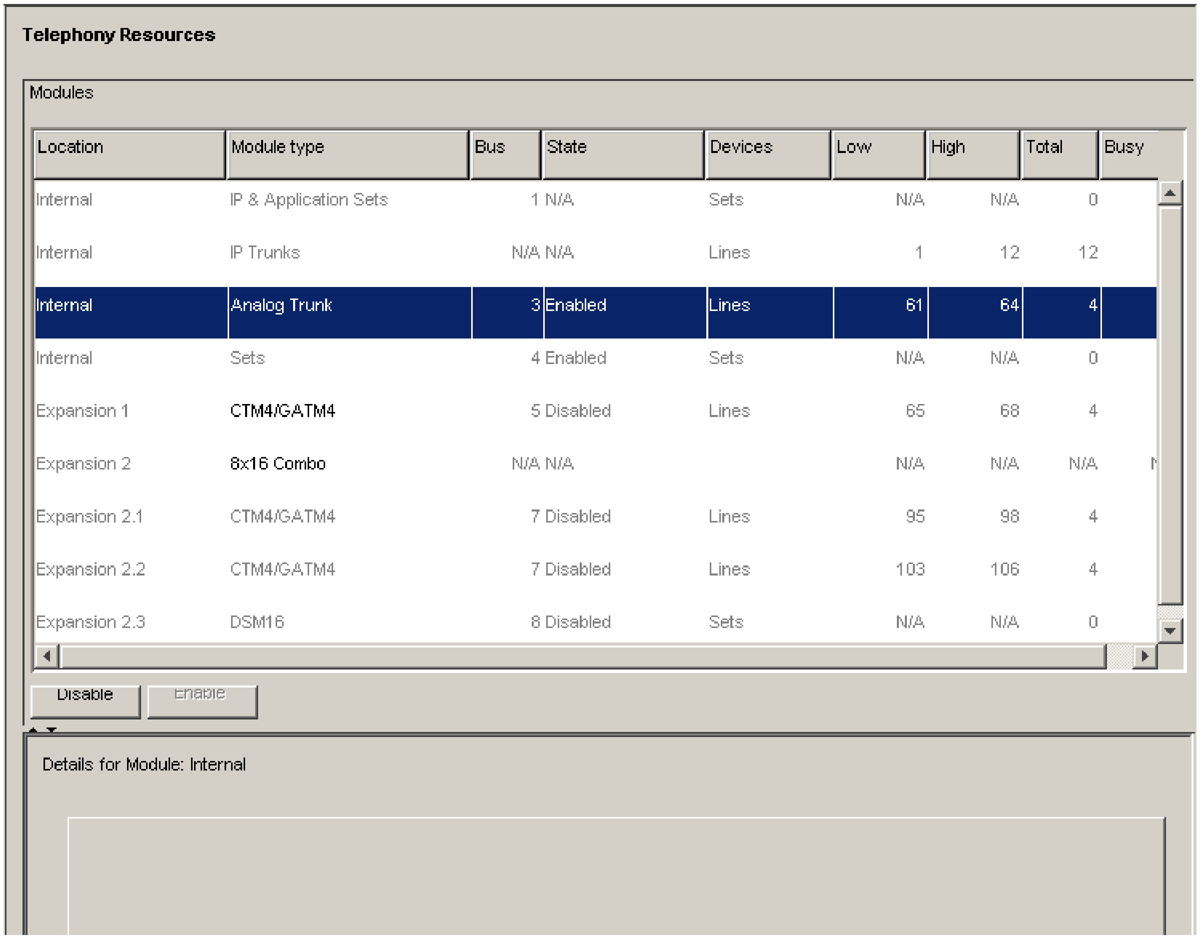
To configure the MBM

1. Open Element Manager and connect to your BCM50 system.
2. Click **Configuration** > **Resources** > **Telephony Resources**.

   The Telephony Resources panel appears (see Figure 27).
3. In the **Location** column of the **Telephony Resources** table, select the location of the MBM that you want to configure.
4. To select the type of MBM that you installed in that location, double-click on the cell of the row you have highlighted to reveal a list and select the MBM type.
5. Click **Enable**.
6. 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 69.

Installing analog devices

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

Documentation describing how to install your analog devices and how to use their features, is supplied with each piece of equipment.

Chapter 4
Installing the analog terminal adapter

This chapter 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 BCM50 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 provides on-premise service only (protected plan wiring only).

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

- "Configuration overview" on page 23
- "Installing the ATA2" on page 24
- "Configuring the ATA2" on page 27

Configuration overview

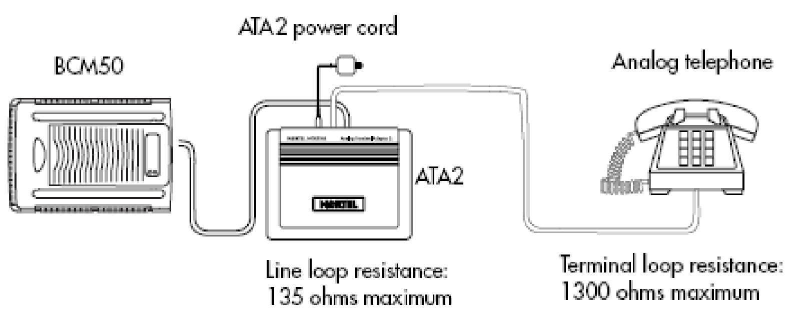
The following sections describe environment configurations for connecting analog and data devices to the main unit using an ATA2:

- "Analog devices" on page 23

Analog devices

Figure 3 on page 23 shows an installation overview for connecting an analog device or analog data device through an ATA2 to the BCM50 main unit.

Figure 3 Analog device installation overview



Installing the ATA2

The following sections provide information on installing the ATA2:

- "Connecting the ATA2" on page 24
- "Mounting the ATA2" on page 25
- "Test insertion loss measurement" on page 25

Connecting the ATA2

After the correct environment has been set up, connect the BCM50 system and the analog device to the ATA2 and then connect the power (see Figure 4).

Figure 4 ATA2 top view

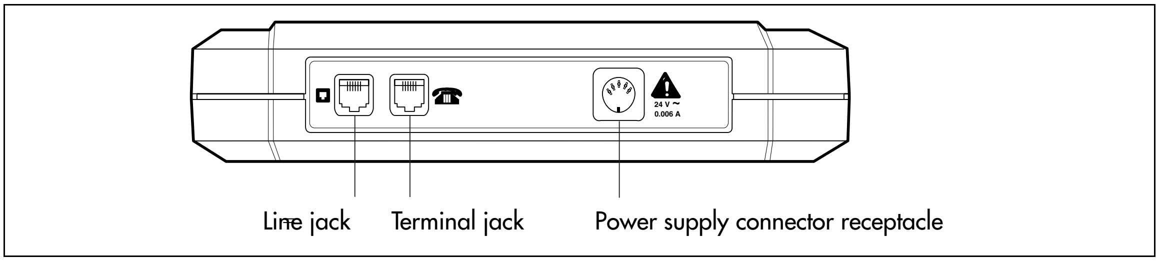
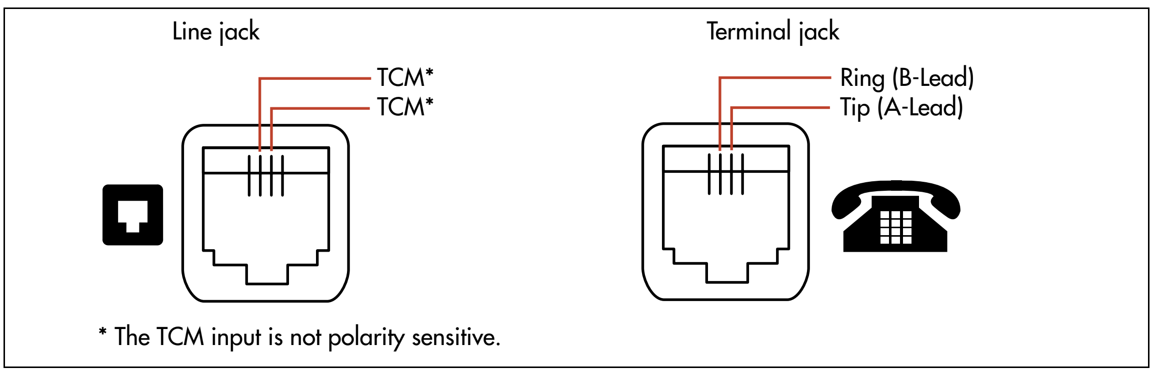


Figure 5 shows the pin-outs for the connection cables.

Figure 5 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 of the line cord to your telephone, modem, or fax machine.
3. Connect one end of a second line cord to the ATA2 line jack.
4. Connect the other end of the line cord to an available station port on the BCM50 system.
5. For a 120-V or 230-V system, plug the DIN connector of the power supply cord into the power supply connector receptacle.

6 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 you have correctly connected the ATA2, you can mount the unit on a wall.

To mount the ATA2 on a wall

1 Select a location for the ATA2 near the BCM50 main unit.

> **Note:** If you are using 0.5 mm wire (24 AWG), select a location within 800 m (2600 ft.) of the BCM50 main unit.

2 Allow 12.5 cm (5 in.) clearance for the line jack, terminal jack, and power supply connector.

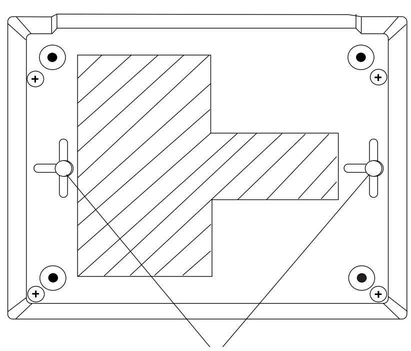
3 Screw two 4-mm (#8) screws into the wall, 130 mm (5-1/4 in.) away from each other. Leave 6 mm (1/4 in.) of the two screws showing.

4 Align the slots at the back of the ATA2 unit over the screws.

5 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 6).

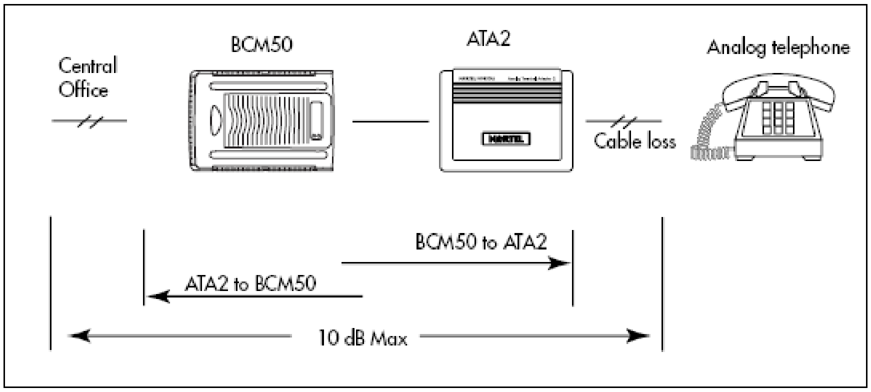
Figure 6 ATA2 back view



Test insertion loss measurement

The maximum loss for and ATA2-to-Central Office (CO) configuration must not exceed 10 dB (see Figure 7 on page 26).

Figure 7 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 *Device Configuration Guide* (NN40020-300).

Chapter 5
ISDN overview

The following provides some general information about using ISDN lines on your BCM50 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:

- "ISDN fundamentals"
- “Services and features for ISDN BRI and PRI” on page 31
- “ISDN hardware” on page 36
- “ISDN standards compatibility” on page 38
- “Planning your ISDN network” on page 39
- “Supported ISDN protocols” on page 40

ISDN fundamentals

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 30
- “ISDN layers” on page 30
- “ISDN bearer capability” on page 31

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 kb/s. 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 kb/s (BRI) and 64 kb/s 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 BCM50. 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 BCM50 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 BCM50 system. Without Layer 2, call processing is not possible.

- Layer 3: Also a logical connection between the ISDN network (your service provider) and the BCM50 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.

These three layers are 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 kb/s, restricted 64 kb/s, or 56 kb/s).

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 33
- “Name and number blocking” on page 33
- “Call by Call Service Selection for PRI” on page 34
- “Emergency 911 dialing” on page 34
- “Two-way DID” on page 35
- “Dialing plan and PRI” on page 35

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 kb/s per loop (depending on the bandwidth supported by your service provider)
- shared digital lines for voice and data ISDN terminal equipment

BCM50 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

BCM50 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 BCM50 sets and ISDN terminal equipment with the capability to show the information).
- Multiline hunt or DN hunting which switches a call to another ISDN line if the line usually used by the Network DN is busy (for 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 (for BRI only).

Transmission of B-channel packet data using nailed up trunks is not supported by BCM50.

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 39 and "Ordering ISDN BRI" on page 39.

The terminal equipment connected to the BCM50 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 BCM50 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 appears as `Private name`, if that is how the service provider has indicated that it mus appear. If the Calling Party Name is unavailable it can appear as `Unknown name`.

Your system can 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

When activated, use **FEATURE 819** to block the outgoing name or number (or both) for each call. Name and number blocking can be used with a BCM50 set.

Note: Name and number blocking is only available in North America.

Consult your customer service representative to determine whether or not this feature is compatible with your provider.

Call by Call Service Selection for PRI

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.

Note: Call by Call Service Selection for PRI is only available in North America.

The service types that may be available, depending on your service provider are

- Public: Public service calls connect your BCM50 set with a Central Office (CO). DID and DOD calls are supported.
- Private: Private service calls connect your BCM50 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 BCM50.
- FX (Foreign Exchange): FX service calls logically connect your BCM50 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

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.

Note: Emergency 911 dialing is only available in North America.

- 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 BCM50 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.

Two-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.

Note: For information on adding integrated lines on an integrated router, refer to the integrated router documentation.

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 BCM50 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 digital BRI ISDN lines are connected to the BCM50b, BCM50ba, and BCM50be main units through the BRI ports (RJ-45) on the front of the main units.

The following describes the hardware:

- "PRI hardware"
- "BRI hardware"

Note: For information on adding integrated lines on an integrated router, refer to the integrated router documentation.

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 cannot have a DTM designated as the PRI primary clock reference is in a network where your BCM50 system is connected back-to-back with another switch using a PRI link. If the other switch is loop-timed to your BCM50 system, your DTM (PRI) can be designated as a timing master.

If your BCM50 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 one of the following:

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

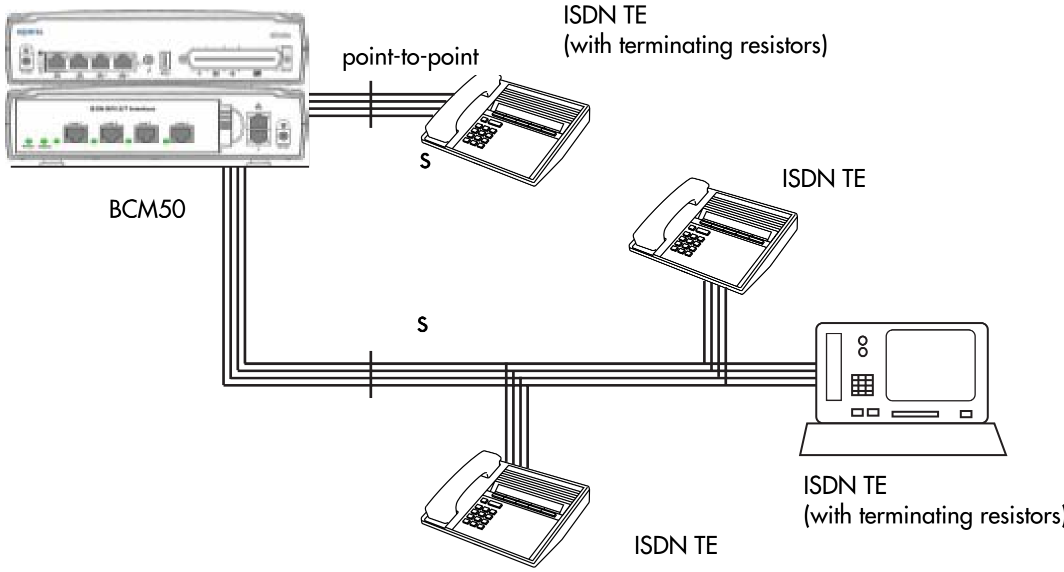
You can add integrated BRI lines on the BCM50a, BCM50e, and BCM50ae main units.

S Reference Point

The S reference point connection provides either a point-to-point or point-to-multipoint digital connection between BCM50 and ISDN terminal equipment (TE) that uses an S interface. Refer to Figure 8.

S loops support up to seven ISDN DNs, which identify TE to the BCM50 system.

Figure 8 S reference point

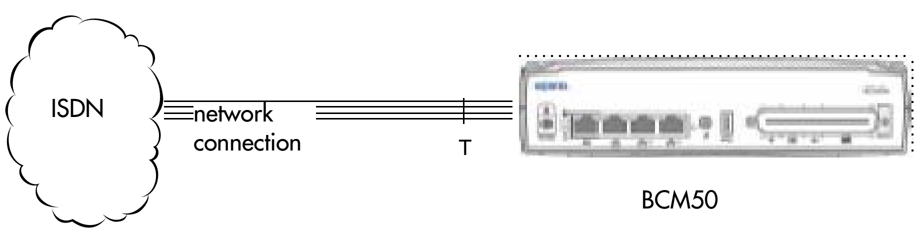


T Reference Points

The T reference point connections provide a point-to-point digital connection between the ISDN network and BCM50. Refer to Figure 9.

A T loop provides lines that can be shared by all BCM50 telephones, peripherals and applications, and ISDN TE.

Figure 9 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 kb/s 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 BCM50 derives timing from the network using T reference points (loops). Terminal equipment on S reference points (loops) derive timing from the BCM50 system.

When you configure the network connections to the BCM50, 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 36.

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 (only in North America). In addition, it manages the maintenance messages travelling between the network and the NT1, and between the NT1 and the BCM50 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, BCM50 ISDN equipment supports National ISDN standards for basic call and calling line identification services. BCM50 BRI is compliant with National ISDN-1 and PRI is compliant with National ISDN-2.

BCM50 does not support EKTS (Electronic Key Telephone System) on PRI.

In Europe, BCM50 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 BCM50, TE and other ISDN equipment.

BCM50 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 BCM50.

Ordering ISDN PRI service in Canada

Ordering ISDN PRI service in the Canada/United States from your service provider. Set the BCM50 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 BCM50 equipment to the Euro ISDN protocol.

Ordering ISDN BRI

The following provides information about how to order ISDN BRI service for your BCM50.

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, you must order it with EKTS turned off. If you are 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 BCM50 system. Contact your service provider for more information about the capability packages it offers. Bellcore/National ISDN Users Forum (NIUF ISDN packages supported by BCM50 (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 BCM50) Basic D-Channel Packet	additional call offering (not supported by BCM50) 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, BCM50 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. BCM50 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 BCM50 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 6
Registering Nortel 20XX and 11XX IP telephones

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

Registering the telephones

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 BCM50 system.

1 Set up the system programming to receive registration under **Resources** > **Telephony Resources**.

a In the Telephony Resources table, click the **IP Sets** row.

The Details for Module panel appears below the Telephony Resources table.

b In the IP Terminal Global Settings tab, select the **Enable registration** check box.

c 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 an alphanumeric password in the **Global password** field.

d 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 42).

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 42, the telephone assumes the parameters it receives from the system, which are described in "Configuring telephone settings" on page 42).

— 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 42, and then follow any of the prompts that appear, as described in "Registering the telephone to the system" on page 42.

— 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 the *Device Configuration Guide* (NN40020-300) for more information on configuring telephones.
- 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 42 to enter the specific network path. "Troubleshooting IP telephones" on page 46 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 Phone 2001/ 2002/2004

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 10 on page 44). These buttons are located directly under the display.

To access the local configuration menu on an IP Phone 2033

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 10 on page 44). These buttons are located directly under the display.

To access the local configuration menu on an IP Phone 2007

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.

To use the IP Phone 2007 dialpad

1. Tap the tool icon (see Figure 10 on page 44).
2. When prompted for a password, using the dialpad enter COLOR*SET (26567*738).

To use the IP Phone 1120E/1140E dialpad

1. Press the **Services** key. See Figure 10 on page 44.
2. Use the navigation keys to find the service to modify.
3. Press **Select.**

Figure 10 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.

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 46.

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 42.

Table 4 on page 45 describes the values for each display parameter.

Table 4 IP telephone server configurations (Sheet 1 of 2)

Field	Value	Description
DHCP	0 or 1	Enter 0 if your network is not using a DHCP server to dispense IP addresses. (Static 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.
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 BCM50 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.

Table 4 IP telephone server configurations (Sheet 2 of 2)

Field	Value	Description
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.
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.		

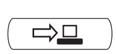
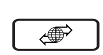
Troubleshooting IP telephones

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

Table 5 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.

To configure an IP Phone 1120E or IP Phone 1140E

There are approximately 45 seconds (s) between plugging in the IP Phone 1120E power adapter and the appearance of the text Nortel. When you see the text Nortel on the phone, you have one s to respond by pressing the four soft keys at the bottom of the display in sequence from left to right, one at a time. If you miss the one-s response time, the IP Phone 1120E attempts to locate the connect server. You can begin the power-up sequence again, or you can double-press the Services key to open the Local diagnostic utilities to access the IP Phone settings.

1 Restart the telephone by disconnecting and reconnecting the power. A splash screen appears with the Nortel logo on it. There is a display interval of 2 to 3 seconds with a 10 second pause.

> **Note:** There is approximately 45 seconds time interval between plugging in the IP Phone 1120E/1140E power adapter and the appearance of the Nortel logo.

2 When the second Nortel text label appears in the middle of the display, immediately press the four soft keys at the bottom of the display in sequence from left to right (see Figure 10 on page 44).
The network configuration menu opens.

> **Note:** You can press the Apply&Reset soft key to save the following settings and to reset the IP Phone.You can press the Exit soft key exit to exit the menu without saving any changes and return to the Network Configuration menu.

3 When the Network Configuration menu opens, the Enable 802.1x (EAP) check box is highlighted. Press **Enter** to toggle this item on and off.
A check mark appears to indicate the item is active.

4 If 802.1x Authentication is enabled, press **Enter** to start the edit mode.

5 Use the keypad to fill in the following information:

- Device ID
- Password (26567*738)
- Retype password

6 Press **Enter**.

> **Note:** If you do not enable 802.1x Authentication, you are not prompted to enter the device ID and password.

7 Use the right navigation key to scroll and highlight Enable 802.1x (LLDP Enable) check box.

8 Press **Enter** to toggle this item on and off.

9 Use the Right navigation key to scroll and highlight DHCP combo box.

10 Press **Enter**.

11 Press the down navigation key to open list box.

12 Configure the following DHCP options:

- **No**—disable DHCP support and enter IP network information manually.
- **Partial**—IP network information (IP address, network mask, and gateway address) are provided by the DHCP server. Enter Server 1 IP address, Server 2 IP address, Port, Action, Retry, and PK numbers manually.
- **Full**—IP network information, Server 1 IP address, Server 2 IP address, and XAS information are provided by the DHCP server.

13 All items are dimmed to prevent manual entry.

> **Note:** A DHCP server and DHCP relay agents must also be installed, configured, and running if you choose Partial DHCP, or Full DHCP configuration.

14 Press **Enter**.

15 Use the right navigation key to scroll and highlight **SET IP**.

16 Press **Enter** to start edit mode.

17 Use the dialpad to fill in the information:

Set IP—a valid IP Phone 1120E IP address

18 Use the right navigation key to scroll and highlight **Gateway**.

19 Press **Enter** to start edit mode.

20 Use the dialpad to fill in the information:

Gateway—the default gateway for the IP Phone 1120E or IP Phone 1140E on the LAN segment to which it is connected

21 Use the right navigation key to scroll and highlight S1 IP.

22 Press **Enter** to start the edit mode.

23 Use the dialpad to fill in the information:

S1 IP—the primary IP address of the BCM50 node IP address for the IP Phone 1120E or IP Phone 1140E

24 Use the right navigation key to scroll and highlight **Port**.

25 Use the dialpad to fill in the information:

S1 Port—a fixed value of 7000

26 Use the right navigation key to scroll and highlight **S1 Action**.

27 Press **Enter** to start edit mode.

28 Use the dialpad to fill in the information. Choose one of the following:

- for TPS only, enter **1**
- for TPS and Secure Media Controller, enter **6** or **1**

29 Use the right navigation key to scroll and highlight **Retry**.

30 Press **Enter** to start edit mode.

31 Use the dialpad to fill in the information.

Retry—the number of times the IP Phone 1120E or IP Phone 1140E attempts to connect to the server

32 Use the right navigation key to scroll and highlight **S1 PK**.

S1 PK—the Private key of the Secure Media Controller to which the IP Phone is connected

33 Press **Enter** to start edit mode.

34 Use the dialpad to fill in the information.

35 Enter **6** or **1**.

36 Press **Enter** to start edit mode.

37 Use the dialpad to fill in the information.

You must enter a 16-digit hexadecimal number. The default is ffffffffffffffff.

38 Use the right navigation key to scroll and highlight **S2 IP**.

39 Press **Enter** to start edit mode.

40 Use the dialpad to fill in the information.

41 **S2 IP**—the secondary BCM50 node IP address for the IP Phone 1120E or IP Phone 1140E

> **Note:** The IP Phone 1120E and IP Phone 1140E can support a primary (S1) and secondary (S1) connect server.

42 Use the right navigation key to scroll and highlight **Port**.

43 Press **Enter** to start edit mode.

44 Use the dialpad to fill in the information.

45 **Port**—same as S1 port

46 Use the right navigation key to scroll and highlight **S2 Action**.

47 Press **Enter** to start edit mode.

48 Use the dialpad to fill in the information. Choose one of the following:

- for TPS only, enter 1
- for TPS and Secure Media Controller, enter **6** or **1**

49 Use the right navigation key to scroll and highlight **Retry**.

50 Press **Enter** to start edit mode.

51 Use the dialpad to fill in the information.

Retry—same as S1

52 Use the right navigation key to scroll and highlight **S2 PK**.

S2 PK—the Private key of the alternate Secure Media Controller to which the IP Phone is connected

53 Press **Enter** to start edit mode.

54 Use the dialpad to fill in the information.

55 Set **S2 PK** to **6** or **1**.

56 Press **Enter** to start edit mode.

57 Use the dialpad to fill in the information.

You must enter a 16-digit hexadecimal number. The default is ffffffffffffffff.

58 Use the right navigation key to scroll and highlight **VoiceVLAN** combo box.

59 Press **Enter**.

60 Press the down navigation key to open the list box.

61 Use the Up/Down navigation keys to scroll and highlight one of the following options:

- No VLAN
- DHCP—VLAN ID is configured automatically to one of the values received from the DHCP server
- LLDP MED—VLAN ID is configured automatically to the value received from 802.1ab LLDP.
- LLDP VLAN Name—VLAN ID is configured automatically to the value received from 802.1ab LLDP.

> **Note:** If LLDP is disabled, LLDP MED and LLDP VLAN Name modes do not appear in the list. If DHCP is disabled, DHCP does not appear in the list.

62 Press **Enter**.

63 Use the right navigation key to scroll and highlight VLAN filter check box.

64 Press **Enter** to toggle this item on and off.

If the VLAN Filter is enabled, packets destined for the IP Phone port are filtered on their MAC address and their VLAN tag. Untagged VLAN packets and tagged VLAN packets that differ from the Telephony VLAN ID are prevented from reaching the IP Phone port. The VLAN Filter check box will appear dimmed if you select No in the VoiceVLAN combo box.

65 Use the right navigation key to scroll and highlight **Disable PC Port.**

66 Press **Enter** to toggle this item on and off.

67 Use the right navigation key to scroll and highlight **DataVLAN** combo box.

68 Press **Enter**.

69 Press the down navigation key to open the list box.

70 Use the up/down navigation keys to scroll and highlight one of the following options:

- No VLAN
- LLDP VLAN Name—VLAN ID is configured automatically to the value received in the VLAN NAME TLV
- VLAN ID value—manual selection of VLAN ID between 1 and 4094 .

> **Note:** If LLDP is disabled, LLDP MED and LLDP VLAN Name modes do not appear in the list.

71 Press **Enter**.

72 Use the right navigation key to scroll and highlight **PC-Port Untag All** check box.

73 Press **Enter** to toggle this item on and off.

If DATA VLAN is enabled, the tag on all traffic destined for the PC port is stripped, by default. To override this action, deselect the **PC-Port Untag All** check box. If DATA VLAN is disabled, the tag on all traffic destined for the PC port is not stripped. To override this action, select the **PC-Port Untag All** check box.

74 Use the right navigation key to scroll and highlight **Duplex** combo box.

75 Press **Enter**.

76 Press the down navigation key to open the list box.

- Auto—Link speed is auto negotiated with the network device and attached PC
- 10BT Full—Link speed is available for up to 10 Megabit Full Duplex on the network and the PC port.
- 100BT Full—Link speed is available for up to 100 Megabit Full Duplex on the network and the PC port.
- 1000BT Full—Link speed is available for up to 1000 Megabit Full Duplex on the network and the PC port.

77 Use the right navigation key to scroll and highlight **Ignore GARP** check box.

78 Press **Enter** to toggle this item on and off.

The GARP feature protects the IP Phone from a Gratuitous ARP Spoof attack from the network.

79 Select **Yes** to enable SRTP media encryption or select **No** to disable media encryption.

The SRTP media encryption feature provides encrypted media. A preshared secret is embedded in the Nortel IP Phone to generate and to exchange encryption parameters without any Call Server involvement.

80 Use the right navigation key to scroll to **Enable PSK SRTP**:

81 If an External Application Server (XAS) is available in the network, use the right navigation key to scroll and highlight the **XAS IP** combo box.

82 Use the dialpad to enter the **XAS IP** address.

The XAS delivers business applications to the IP Phone.

83 If the XAS supports graphical displays, use the right navigation key to scroll and highlight Graphical XAS check box.

84 Press **Enter** to toggle this item on and off.

85 Use the right navigation key to scroll and highlight **Port** combo box.

86 Press **Enter** to start edit mode.

87 Use the dialpad to fill in the information.

88 Upgrade the IP Phone 1120E or IP Phone 1140E firmware.

The IP Phone 1120E supports remote firmware upgrades through a TFTP process and an automated UFTP process.

89 The method to upgrade the firmware depends on the version of BCM software you are running.

90 Check for dial tone and the correct DN on the display.

Operation issues

Table 6 provides solutions to potential problems.

Table 6 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 4 on page 45.
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 BCM50, 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 **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.

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

To deregister a IP telephone from the IP record

1. Navigate to **Configuration** > **Telephony** > **Sets** > **Active Sets**.
2. In the Active Sets table, click the **Capabilities and Preferences** tab.
3. Select the DN you want to deregister and click the **IP Terminal Details** tab under the DN.
4. Click **Deregister DN**.
5. Reregister the telephone, as described in "Registering the telephones" on page 41.

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

Chapter 7
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 BCM50 IP telephony and Voice over IP (VoIP) features:

- "IP telephones and VoIP trunks" on page 56
- "Creating the IP telephony network" on page 57
- "Key IP telephony concepts" on page 60

BCM50 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 BCM50 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"

IP telephones

IP telephones offer the functionality of regular telephones, but do not require a hardwire connection to the BCM50. Instead, they must be plugged into an IP network which is connected to the through the integrated interface (LAN card) on the BCM50.

Calls made from IP telephones through the BCM50 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 IP Phone series of telephones, or are accessed through your desktop or laptop computer, as in the case of the IP Software Phone 2050.

VoIP trunks

VoIP trunks allow voice signals to travel across IP networks. A gateway within the BCM50 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.

This system supports SIP trunks and H.323 trunks. Both types of trunks support connections to other BCMs, a central call server such as Succession 1000/M, and trunk-based applications. SIP trunks and H.323 trunks are assigned to a single Pool, and the routing decision to route calls via H.323 or SIP is made based on the routing modes of the two services (Direct/Gatekeeper/Proxy) and the combined routing table.

Creating the IP telephony network

The following explains the components of the BCM50 system and the devices it interoperates to create a network.

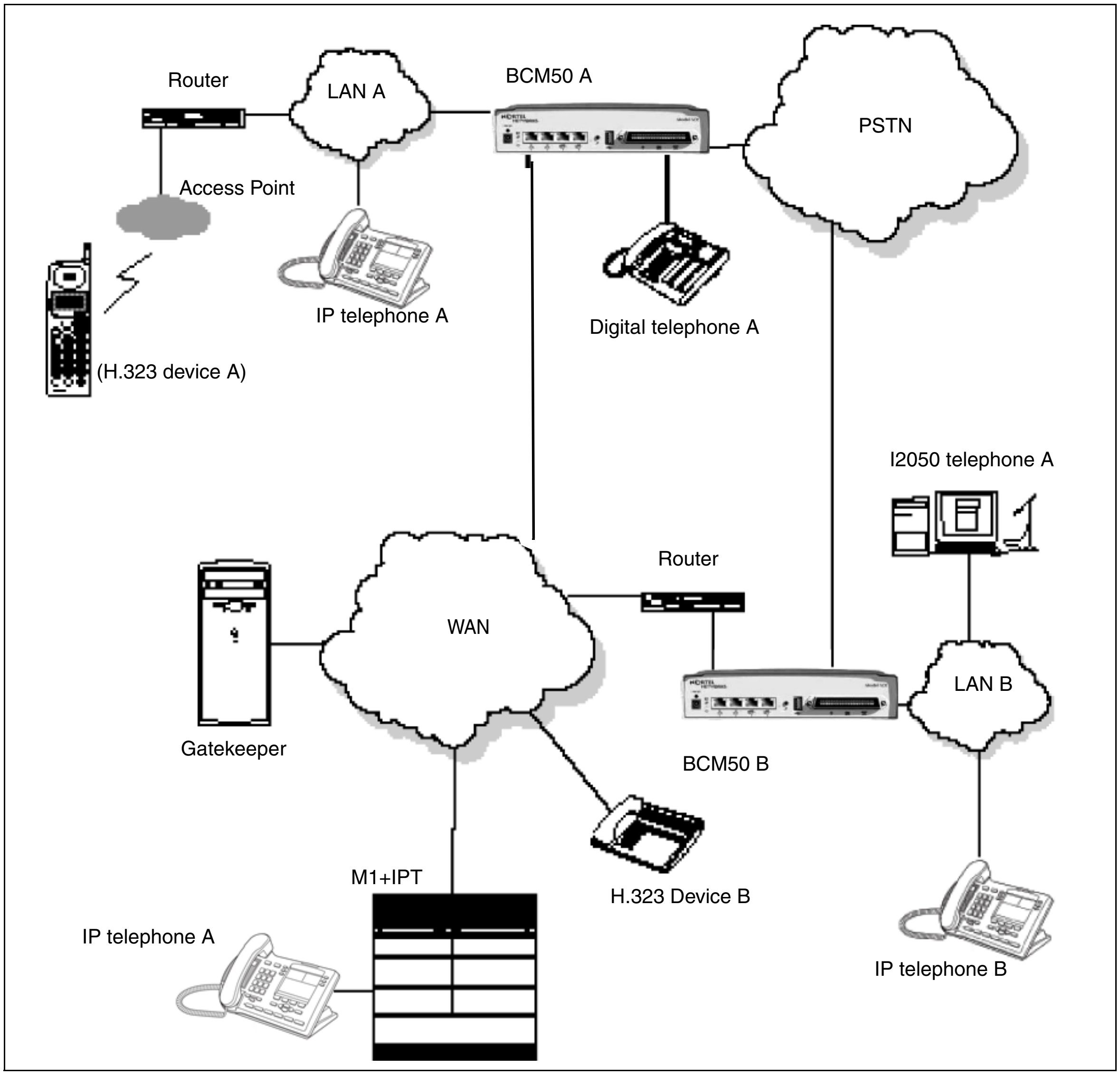
The information under the following headings describes the various components of the system:

- "M1-IPT" on page 59
- "Telephones" on page 59
- "Gatekeepers on the network" on page 59
- "IP network" on page 60
- "Public Switched Telephone Network" on page 60

Figure 11 shows components of a BCM50 network configuration.

In this example, two BCM50 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 BCM50 systems use VoIP trunks through a common IP network to connect to the Meridian (M1-IPT) system.

Figure 11 Network diagram



Networking with BCM50

The BCM50 is a key building block in creating your communications network. It interoperates with many devices, including the Meridian 1 system, and H.323 and SIP devices. The BCM50 system can be connected to devices through multiple IP networks, as well as through the PSTN. Multiple BCM50 systems also can be linked together on a network of VoIP trunks and/or dedicated physical lines.

The BCM50 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 BCM50 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 BCM50 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 BCM50s 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 BCM50 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 BCM50 can communicate using digital telephones (Model 7000, 7100, 7208, 7316, 7316E/7316E+KIMs, 7406 and 7406E (cordless telephones), Norstar M-series telephones, ISDN telephones, analog telephones, and IP telephones and applications. With this much flexibility, the BCM50 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 BCM50 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 BCM50 to voice channels.

The IP telephones connect to the BCM50 across an IP network through either a LAN or a WAN. From the BCM50 connection, they can then use standard lines or VoIP trunks to communicate to other telephones on other public or private networks. The BCM50 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 BCM50 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 BCM50 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 BCM50 B over the network. BCM50 B then passes the call to IP telephone B.

The BCM50 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 BCM50 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 BCM50 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 BCM50, a WAN is any IP network connected to a WAN card on the BCM50 system. This may also be a direct connection to another BCM50 system.

If you want to deploy IP telephones that will be connected to a LAN outside of the LAN that the BCM50 is installed on, you must ensure the BCM50 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 BCM50, a LAN is any IP network connected to the integrated interface (a LAN card) on the BCM50 system. Often, the LAN can include a router that forms a connection to the Internet. A BCM50 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 BCM50 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 61
- "Jitter buffer" on page 61
- "QoS routing" on page 62

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 (kb/s) while G.729 samples at a far lower rate of 8 kb/s.

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 BCM50, or if Keep DN Alive is enabled for an offline telephone.

The BCM50 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 BCM50, 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 BCM50 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 BCM50 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 BCM50 assigns Expedited Forwarding (EF) PHB for voice media packets. On the BCM50, these assignments cannot be adjusted.

Chapter 8
Relocating telephones

This chapter 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 63
- "Moving IP telephones" on page 64
- "" on page 65 provides a list of the user cards for individual types of telephones

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 64.

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 to relocate digital telephones

1. In the Element Manager, go to **Configuration** > **Telephony** > **Global Settings** > **Feature Settings**.
2. In the **Feature Settings** area, select the **Set relocation** check box.
3. Move the telephone by physically unplugging the telephone and plugging it in again at another location.

   It can take up to 45 seconds for the system to recognize the telephone.
4. Clear the **Set relocation** check box.

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 IP Software Phone 2050 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.
3. Click **IP Terminal** details.
4. Select the **Keep DN alive** check box.

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

Moving IP telephones

IP telephones retain their DN when you move them 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 three-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 three-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.

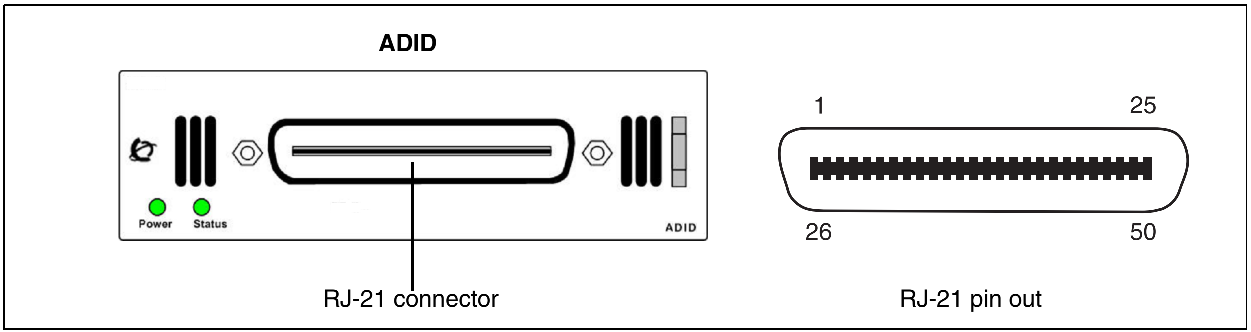
2 Disconnect the network connection and the power connection from the telephone.

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

Appendix A
ADID wiring chart

Analog telephone lines connect to the ADID4 or ADID8 through the RJ-21 connector on the front of the media bay module (MBM). See the figure ADID RJ-21 connector (page 67).

Figure 12   ADID RJ-21 connector



The table ADID4 and ADID8 RJ-21 connector wiring (page 67) lists the wiring details for the RJ-21 connector on the ADID4 and ADID8. Use the first four lines for the ADID4 and use all eight lines for the ADID8.

Table 7   ADID4 and ADID8 RJ-21 connector wiring

Line	Pin	Connection	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
1	26	Tip	**065**	**095**
	1	Ring		
2	27	Tip	**066**	**096**
	2	Ring		
3	28	Tip	**067**	**097**
	3	Ring		
4	29	Tip	**068**	**098**
	4	Ring		
5	30	Tip	**073**	**103**
	5	Ring		
6	31	Tip	**074**	**104**
	6	Ring		
7	32	Tip	**075**	**105**
	7	Ring		
8	33	Tip	**076**	**106**
	8	Ring		

Appendix B
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

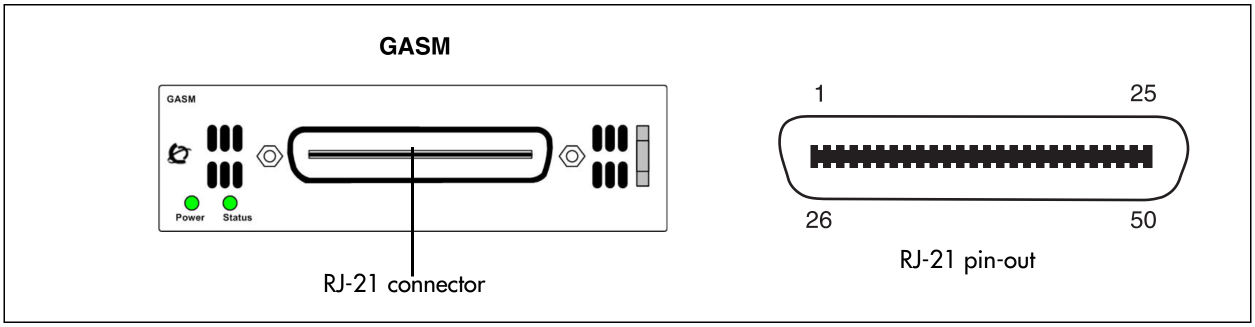


Table 8 lists the wiring details for the RJ-21 connector on the ASM.

Table 8   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 8 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 C
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

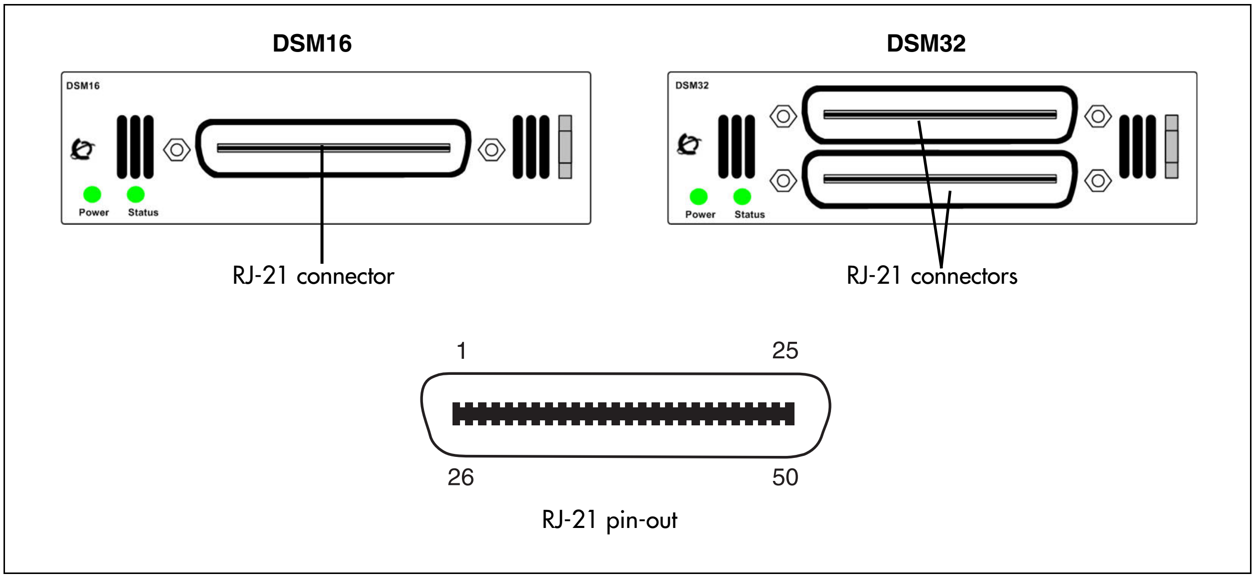


Table 9 lists the wiring details for the RJ-21 connectors on the DSM16 and DSM32.

Table 9 DSM16 and DSM32 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	
				DSM16 or Lower DSM32 RJ-21	Upper DSM32 RJ-21	DSM16 or Lower DSM32 RJ-21	Upper DSM32 RJ-21
1	26	Tip	White-Blue	237	253	269	285
	1	Ring	Blue-White				
2	27	Tip	White-Orange	238	254	270	286
	2	Ring	Orange-White				
3	28	Tip	White-Green	239	255	271	287
	3	Ring	Green-White				
4	29	Tip	White-Brown	240	256	272	288
	4	Ring	Brown-White				

Table 9 DSM16 and DSM32 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	
				DSM16 or Lower DSM32 RJ-21	Upper DSM32 RJ-21	DSM16 or Lower DSM32 RJ-21	Upper DSM32 RJ-21
5	30	Tip	White-Slate	**241**	**257**	**273**	**289**
	5	Ring	Slate-White				
6	31	Tip	Red-Blue	**242**	**258**	**274**	**290**
	6	Ring	Blue-Red				
7	32	Tip	Red-Orange	**243**	**259**	**275**	**291**
	7	Ring	Orange-Red				
8	33	Tip	Red-Green	**244**	**260**	**276**	**292**
	8	Ring	Green-Red				
9	34	Tip	Red-Brown	**245**	**261**	**277**	**293**
	9	Ring	Brown-Red				
10	35	Tip	Red-Slate	**246**	**262**	**278**	**294**
	10	Ring	Slate-Red				
11	36	Tip	Black-Blue	**247**	**263**	**279**	**295**
	11	Ring	Blue-Black				
12	37	Tip	Black-Orange	**248**	**264**	**280**	**296**
	12	Ring	Orange-Black				
13	38	Tip	Black-Green	**249**	**265**	**281**	**297**
	13	Ring	Green-Black				
14	39	Tip	Black-Brown	**250**	**266**	**282**	**298**
	14	Ring	Brown-Black				
15	40	Tip	Black-Slate	**251**	**267**	**283**	**299**
	15	Ring	Slate-Black				
16	41	Tip	Yellow-Blue	**252**	**268**	**284**	**300**
	16	Ring	Blue-Yellow				
—	42	No connection	Yellow-Orange	—	—	—	—
	17	No connection	Orange-Yellow				
...	...	...	...	...	...	...	...
—	50	No connection	Violet-Slate	—	—	—	—
	25	No connection	Slate-Violet				

Appendix D
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

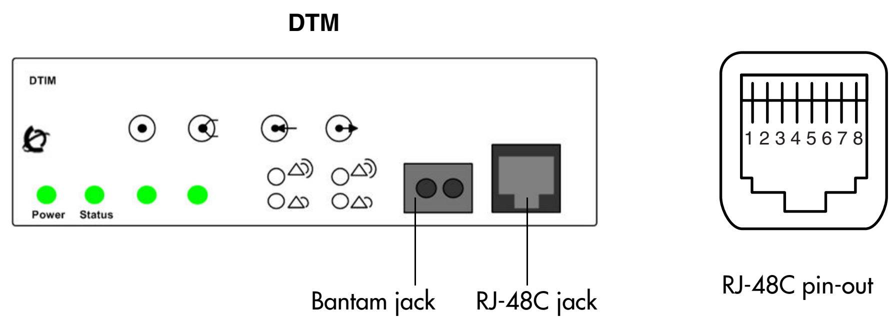


Table 10 and Table 11 list the wiring details for the RJ-48C port.

Table 10   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 11   DTM line numbering

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

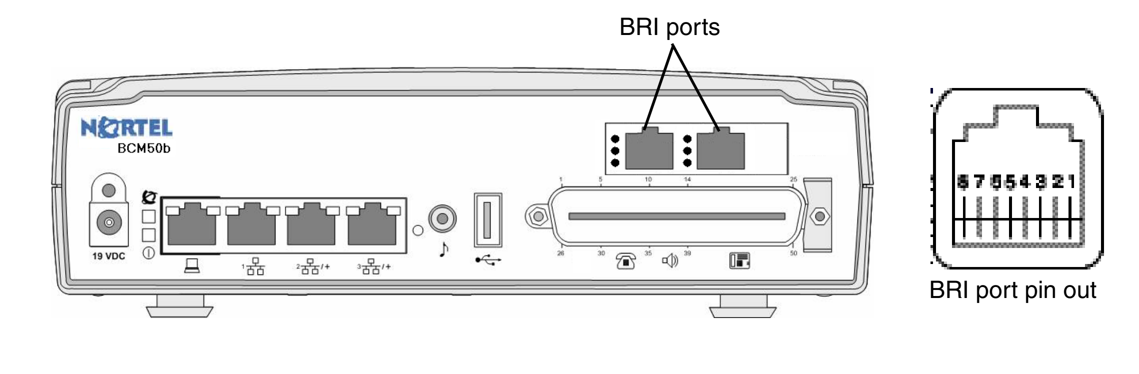
Appendix E
BRI wiring chart

The digital BRI ISDN lines are connected to the BCM50b, BCM50ba, and BCM50be main units through the BRI ports (RJ-45) on the front of the main units. See the figure "BRI ports and pin out (BCM50b shown)" on page 75.

The figure "BRI ports and pin out (BCM50b shown)" on page 75, the table BRI port wiring (page 75), and the table "BRI line numbering" on page 76 apply to S-Loop and T-Loop connections. S-Loop are used to connect S-Loop devices such as video phones, terminal adapters, and Grp 3 Fax machines. The T-Loops are used to connect to the CO/PSTN.

Warning: For a U-loop connection, the BRI port 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 BRI ports and pin out (BCM50b shown)



The table "BRI port wiring" on page 75 and the table "BRI line numbering" on page 76 list the wiring details for the RJ-45 ports.

Table 12 BRI 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 13 BRI 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

Appendix F
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 17). You can connect up to four BRI ISDN lines to the BRIM.

Figure 17, Table 14, and Table 15 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 17 BRIM RJ-45 ports

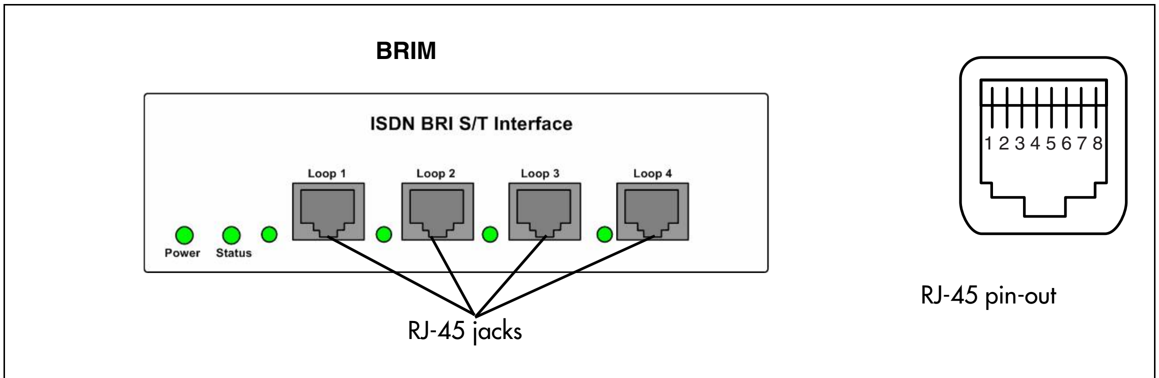


Table 14 and Table 15 list the wiring details for the RJ-45 ports.

Table 14 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 15 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

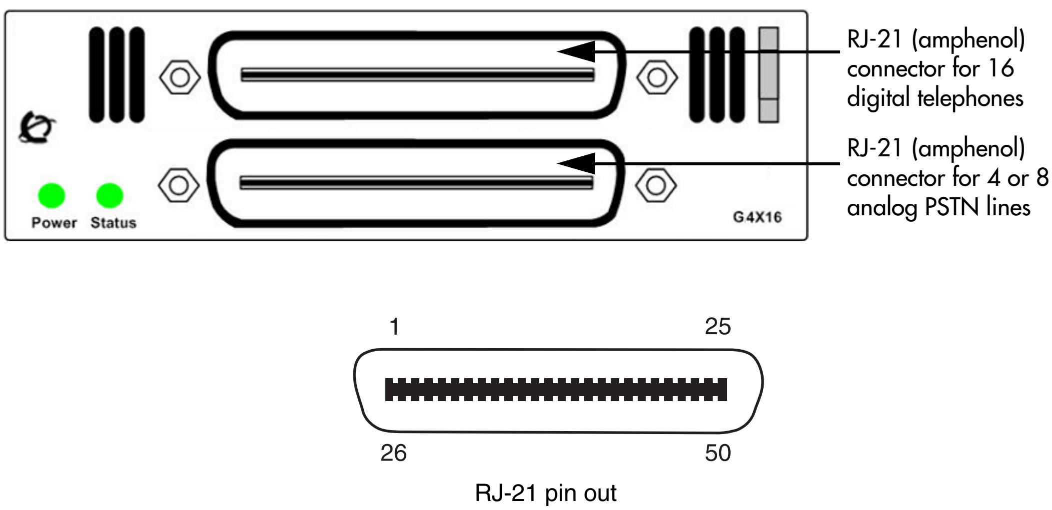
Appendix G
G4x16 and G8x16 wiring charts

You can connect 4 or 8 analog telephone lines and up to 16 digital telephones to the G4x16 or G8x16 media bay module (MBM).

The analog telephones lines are connected to the G4x16 or G8x16 through the lower RJ-21 (amphenol) connector on the front of the MBM. See the figure G4x16/G8x16 connectors (page 79).

The digital telephones, such as the Business Series Telephones, are connected to the upper RJ-21 (amphenol) connector on the front of the G4x16 or G8x16 MBM.

Figure 18 G4x16/G8x16 connectors



The table G4x16/G8x16 MBM upper RJ-21 (amphenol) connector wiring for digital telephones (page 79) lists the wiring details for the upper RJ-21 (amphenol) connector, and the table G4x16/G8x16 lower RJ-21 (amphenol) connector wiring for analog PSTN lines (page 81) lists the wiring details for the lower RJ-21 (amphenol) connector on the G4x16 and G8x16 MBMs.

Table 16 G4x16/G8x16 MBM upper RJ-21 (amphenol) connector wiring for digital telephones (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	**253**	**285**
	1	Ring	Blue-White		
2	27	Tip	White-Orange	**254**	**286**
	2	Ring	Orange-White		

Table 16 G4x16/G8x16 MBM upper RJ-21 (amphenol) connector wiring for digital telephones (Sheet 2 of 2)

Set	Pin	Connection	Wire color	Default DN on Expansion port 1	Default DN on Expansion port 2
3	28	Tip	White-Green	**255**	**287**
	3	Ring	Green-White		
4	29	Tip	White-Brown	**256**	**288**
	4	Ring	Brown-White		
5	30	Tip	White-Slate	**257**	**289**
	5	Ring	Slate-White		
6	31	Tip	Red-Blue	**258**	**290**
	6	Ring	Blue-Red		
7	32	Tip	Red-Orange	**259**	**291**
	7	Ring	Orange-Red		
8	33	Tip	Red-Green	**260**	**292**
	8	Ring	Green-Red		
9	34	Tip	Red-Brown	**261**	**293**
	9	Ring	Brown-Red		
10	35	Tip	Red-Slate	**262**	**294**
	10	Ring	Slate-Red		
11	36	Tip	Black-Blue	**263**	**295**
	11	Ring	Blue-Black		
12	37	Tip	Black-Orange	**264**	**296**
	12	Ring	Orange-Black		
13	38	Tip	Black-Green	**265**	**297**
	13	Ring	Green-Black		
14	39	Tip	Black-Brown	**266**	**298**
	14	Ring	Brown-Black		
15	40	Tip	Black-Slate	**267**	**299**
	15	Ring	Slate-Black		
16	41	Tip	Yellow-Blue	**268**	**300**
	16	Ring	Blue-Yellow		
—	42	No connection	Yellow-Orange	**—**	**—**
	17	No connection	Orange-Yellow		
⋮	⋮	⋮	⋮	⋮	⋮
—	50	No connection	Violet-Slate	**—**	**—**
	25	No connection	Slate-Violet		

Table 17 G4x16/G8x16 lower RJ-21 (amphenol) connector wiring for analog PSTN lines (Sheet 1 of 2)

Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
1	26	Tip	White-Blue	**065**	**095**
	1	Ring	Blue-White		
2	27	Tip	White-Orange	**066**	**096**
	2	Ring	Orange-White		
—	28	No connection	White-Green	**—**	**—**
	3	No connection	Green-White		
—	29	No connection	White-Brown	**—**	**—**
	4	No connection	Brown-White		
3	30	Tip	White-Slate	**067**	**097**
	5	Ring	Slate-White		
4	31	Tip	Red-Blue	**068**	**098**
	6	Ring	Blue-Red		
—	32	No connection	Red-Orange	**—**	**—**
	7	No connection	Orange-Red		
The following part of the wiring chart applies only to the G8x16 MBM.					
—	33	No connection	Red-Green	**—**	**—**
	8	No connection	Green-Red		
5	34	Tip	Red-Brown	**073**	**103**
	9	Ring	Brown-Red		
6	35	Tip	Red-Slate	**074**	**104**
	10	Ring	Slate-Red		
—	36	No connection	Black-Blue	**—**	**—**
	11	No connection	Blue-Black		
—	37	No connection	Black-Orange	**—**	**—**
	12	No connection	Orange-Black		
7	38	Tip	Black-Green	**075**	**105**
	13	Ring	Green-Black		
8	39	Tip	Black-Brown	**076**	**106**
	14	Ring	Brown-Black		
—	40	No connection	Black-Slate	**—**	**—**
	15	No connection	Slate-Black		
The following part of the wiring chart applies to both the G4x16 and G8x16 MBMs.					
⋮	⋮	⋮	⋮	⋮	⋮
—	49	No connection	Violet-Brown	**—**	**—**
	24	No connection	Brown-Violet		

Table 17 G4x16/G8x16 lower RJ-21 (amphenol) connector wiring for analog PSTN lines (Sheet 2 of 2)

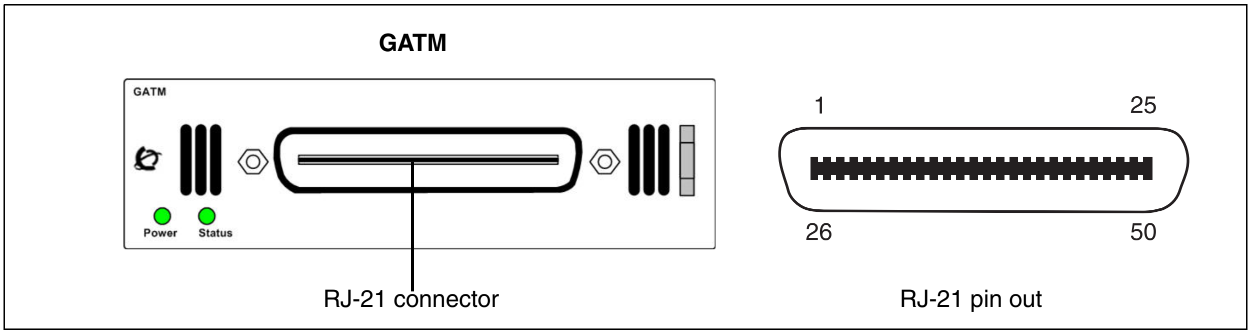
Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
Aux (see Note)	50	Tip	Violet-Slate	—	—
	25	Ring	Slate-Violet		

Note: The Aux port supports full data speeds. When the line is in use by an analog device, the icon is lit on the phone. If you try to seize the line using the phone, the display shows “in use.” Also, in the event of a power failure, an analog set on line 1 goes active (powered by the CO).

Appendix H
GATM wiring chart

Analog telephone lines connect to the GATM4 or GATM8 through the RJ-21 connector on the front of the media bay module (MBM). See the figure GATM RJ-21 connector (page 83).

Figure 19   GATM RJ-21 connector



The table GATM4 RJ-21 connector wiring (page 83) lists the wiring details for the RJ-21 connector on the GATM4.

Table 18   GATM4 RJ-21 connector wiring (Sheet 1 of 2)

Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
1	26	Tip	White-Blue	**065**	**095**
	1	Ring	Blue-White		
2	27	Tip	White-Orange	**066**	**096**
	2	Ring	Orange-White		
—	28	No connection	White-Green	—	—
	3	No connection	Green-White		
—	29	No connection	White-Brown	—	—
	4	No connection	Brown-White		
3	30	Tip	White-Slate	**067**	**097**
	5	Ring	Slate-White		
4	31	Tip	Red-Blue	**068**	**098**
	6	Ring	Blue-Red		
—	32	No connection	Red-Orange	—	—
	7	No connection	Orange-Red		
⋮	⋮	⋮	⋮	⋮	⋮

Table 18 GATM4 RJ-21 connector wiring (Sheet 2 of 2)

Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
—	49	No connection	Violet-Brown	—	—
	24	No connection	Brown-Violet		
Aux	50	Tip	Violet-Slate	—	—
	25	Ring	Slate-Violet		
Note: The AUX port supports full data speeds. When the line is in use by an analog device, the icon is lit on the phone to indicate it is in use. If you try to seize the line using the phone, the display shows “in use”. Also, if a power failure occurs, an analog set on line 1 activates (powered by the CO).					

The table GATM8 RJ-21 connector wiring (page 84) lists the wiring details for the RJ-21 connector on the GATM8.

Table 19 GATM8 RJ-21 connector wiring (Sheet 1 of 2)

Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
1	26	Tip	White-Blue	**065**	**095**
	1	Ring	Blue-White		
2	27	Tip	White-Orange	**066**	**096**
	2	Ring	Orange-White		
—	28	No connection	White-Green	—	—
	3	No connection	Green-White		
—	29	No connection	White-Brown	—	—
	4	No connection	Brown-White		
3	30	Tip	White-Slate	**067**	**097**
	5	Ring	Slate-White		
4	31	Tip	Red-Blue	**068**	**098**
	6	Ring	Blue-Red		
—	32	No connection	Red-Orange	—	—
	7	No connection	Orange-Red		
—	33	No connection	Red-Green	—	—
	8	No connection	Green-Red		
5	34	Tip	Red-Brown	**073**	**103**
	9	Ring	Brown-Red		
6	35	Tip	Red-Slate	**074**	**104**
	10	Ring	Slate-Red		
—	36	No connection	Black-Blue	—	—
	11	No connection	Blue-Black		
—	37	No connection	Black-Orange	—	—
	12	No connection	Orange-Black		

Table 19 GATM8 RJ-21 connector wiring (Sheet 2 of 2)

Line	Pin	Connection	Wire color	Default line numbers on Expansion port 1	Default line numbers on Expansion port 2
7	38	Tip	Black-Green	**075**	**105**
	13	Ring	Green-Black		
8	39	Tip	Black-Brown	**076**	**106**
	14	Ring	Brown-Black		
—	40	No connection	Black-Slate	—	—
	15	No connection	Slate-Black		
. . .	. . .	. . .	. . .	. . .	. . .
—	49	No connection	Violet-Brown	—	—
	24	No connection	Brown-Violet		
Aux (see Note)	50	Tip	Violet-Slate	—	—
	25	Ring	Slate-Violet		

Note: The AUX port supports full data speeds. When the line is in use by an analog device, the icon is lit on the phone to indicate it is in use. If you try to seize the line using the phone, the display shows “in use”. Also, if a power failure occurs, an analog set on line 1 activates (powered by the CO).

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