Nortel Business Communications Manager 450 1.0

Installation—Devices

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Contents

New in this release

This is the initial release of the BCM450 platform. This document contains information about devices supported by the BCM450 system in Release 1.0.

Features

See the following sections for information found in this guide.

- The Overview of key hardware elements and devices section includes information about BCM450 media bay modules, applications, digital devices, IP devices and wireless devices.
- The Device compatibility and installation requirements section includes information about using Integrated Services Digital Network (ISDN) lines on your BCM450 system.
- The IP phones and IP telephony section includes an overview of the components that comprise the BCM450 IP telephony and Voice over IP (VoIP) features
- The Media bay module configuration section includes information about installing MBMs.
- The IP phone registration section includes information about registering your IP phone.
- The IP Phone 1120E or IP Phone 1140E configuration section includes procedures about configuring your IP Phone 1120E or 1140E.
- The Telephone relocation section includes procedures about moving a telephone within the system so that the telephone programming follows the telephone to the new location.
- The Media encryption on IP phones section includes procedures about enabling media encryption on your IP phone.

Introduction

This document contains conceptual, task-based, and reference information about analog, digital, IP, and Integrated Services Digital Network (ISDN) devices on a Business Communications Manager 450 (BCM) system.

The information in this guide explains how to

- install and configure components
- register and relocate telephones and devices
- enable media encryption

Use Element Manager, Startup Profile, and Telset Administration to configure BCM450 parameters.

Navigation

Overview of key hardware elements and devices

The Business Communications Manager 450 system provides private network and telephony management capability to small and medium-sized businesses.

The BCM450 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

This section describes the telephony devices BCM450 supports.

Navigation

BCM450 main and expansion units

BCM450 includes the following key elements:

- BCM450 main unit
- BCM450 expansion unit

Media bay module types

BCM450 includes the following media bay modules (MBM):

- 4 x 16
- ADID4
- ADID8
- ASM8, ASM8+
- BRIM
- CTM4, CTM8
- DSM16, DSM32
- DSM16+, DSM32+
- DTM
- FEM
- GASM
- GATM4, GATM8
- G4 x 16
- G8 x 16
- R2MFC

BCM450 features

BCM450 supports the complete range of IP telephony features offered by existing BCM products.

BCM450 applications

BCM450 supports many applications provided on the existing BCM platform.

Enter the appropriate keycodes to enable the following features (no additional hardware required):

- Voice Messaging for standard voice mail and auto-attendant features
- Unified Messaging to provide integrated voice mail management between voice mail and common e-mail applications
- Fax Suite to providing support for attached analog fax devices
- Voice networking features
- LAN CTE (computer telephony engine)
- VEWAN
- IP Music

- Contact Center

Digital devices

BCM450 supports corded display sets, cordless sets, key indicator modules (KIM), the Business Series Terminal (BST) Doorphone, and the Central Answering Position.

Navigation

- Corded display sets and options (page 13)
- Cordless sets and options (page 13)
- Key indicator module (page 13)
- BST Doorphone (page 14)

Corded display sets and options

- **T7000 (International only):** four memory buttons, without display or indicators.
- **T7100**: one-line display, and one memory button without indicator.
- **T7208**: one-line display, and eight memory buttons with indicators.
- **T7316**: two-line display, three display buttons, 16 memory buttons with indicators, and 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, and eight memory buttons without indicators; handsfree, mute, and headset buttons (under the dial pad).

Cordless sets and options

- **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

The KIM includes 24 memory buttons with indicators.

BST Doorphone

Use BST Doorphone as an intercom to control access to your building. Press the Call button on the BST Doorphone to call one or more telephones, or 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

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 KIMs.

Wireless devices

BCM450 supports Digital Enhanced Cordless Telecommunications (DECT) handsets, digital mobility phones, and WLAN handsets.

DECT 4xxx series handsets

- DECT 413x series handsets: three display soft keys, four-line handset display, and text messaging.
- DECT 414x series handsets: three display soft keys, four-line handset display, loudspeaker capability, and text messaging.

Digital mobility phones

- Digital Mobility Phone 7420: three display soft keys, four-line handset display.
- Digital Mobility Phone 7430: three display soft keys, and four-line handset display, text messaging.
- Digital Mobility Phone 7440: three display soft keys, and four-line handset display, loudspeaker capability, and text messaging.

WLAN handsets 221X

WLAN handsets use VoIP technology and Push-to-Talk, which enables two-way communication with another BCM user.

The handsets communicate with the BCM system and with the WLAN IP Telephony Manager 2245. 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 interoperate 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.

WLAN Handsets 6120 and 6140

The WLAN Handsets 6120 and 6140 operate with the BCM and the WLAN IP Telephony Manager 2245. They are fully functional handsets specifically designed for the busy office environment. The WLAN Handsets 6120 and 6140 use radio wave technology to send and receive voice and data transmissions. They operate much like a cell phone. However, the Handsets use the private communication system installed in your facility and will not operate outside the area covered by this system.

IP devices

BCM450 supports IP phones, IP phone expansion modules, a softphone, and an audio conference phone.

Navigation

- IP phone registration (page 15)
- IP phone configuration (page 16)
- Registering the telephone to the system (page 17)
- Configuring telephone settings (page 17)
- Troubleshooting IP telephones (page 19)
- 1100 series IP phones (page 20)
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- IP Key Expansion Module (page 21)
- IP Softphone 2050 (page 21)
- IP Audio Conference Phone 2033 (page 22)

IP phone registration

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

- Set up the system programming to receive registration under Resources > Telephony Resources. For more information see, IP phone registration (page 53)
- Configure each telephone.

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.

Attention: If the telephone displays a prompt that indicates it cannot find the server, follow the instructions in IP phone registration (page 53) to enter the specific network path. Troubleshooting IP telephones (page 19) describes other possible prompt messages.

If you do not automatically register to the system, you can configure the telephone settings to enable you to access a system on the network. You must perform additional steps if your IP telephone does not connect to the same LAN to which the system connects.

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 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, and the six keys at the top of the display become labelled.

If you experience problems with IP telephone registration, see Troubleshooting IP telephones (page 19).

If the DN record is not configured, as with auto-assigned DNs, you can only place local calls until other lines assigned in the DN record.

If no one registered the telephone previously, you receive a “New Set” message. Enter the information as prompted. See Registering the telephone to the system (page 17).

IP phone configuration

The configuration of the IP phones depends on whether you use Dynamic Host Control Protocol (DHCP) on the system.

- If you use DHCP service on the system, or you configured the Customer DHCP server to hand out 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 (page 17)the telephone assumes the parameters it receives from the system, which are described in Configuring telephone settings (page 17)

- If you did not configure DHCP to provide system information, or if you do not use 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 (page 17), and then

follow the prompts that appear, as described in Registering the telephone to the system (page 17)

- If an external DHCP server does not exist, 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. For more information see the *BCM450 1.0 Configuration–Devices Guide* (NN40160-500)
- 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.

If the telephone displays a prompt that indicates it cannot find the server, follow the instructions in Configuring telephone settings (page 17)to enter the specific network path.Troubleshooting IP telephones (page 19) 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. For more information about accessing the telephone configuration menus see, IP phone registration (page 53)

If you experience problems with IP telephone registration, refer to the section Troubleshooting IP telephones (page 19).

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. The following table describes the values for each display parameter. For more information see, Registering the telephone to the system (page 17)

Table 1 IP telephone server configurations

Field	Value	Description
DHCP	0 or 1	Enter 0 if your network does not use a DHCP server to dispense IP addresses. (Static DHCP) Enter 1 if your network uses a DHCP server.
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 connects.
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 connects, (for example, a direct HUB connection), enter the Published IP address of the BCM450 as the DEF GW. If the IP telephone does not connect directly to the Published IP address network adapter, set the DEF GW to the IP address of the network adaptor to which the telephone connects.
Emulation Key Mapping	0 or 1	0 = Handset 1= Handsfree Default setting is 1 (handsfree)—do not change. 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 BCM 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>	Configure this to the number of times you want the telephone to retry the connection to the system.

Table 1 IP telephone server configurations

Field	Value	Description
S2 IP	<IP address>	This is the Published IP address of the second BCM system to which you want to register the telephone. It can match the S1 setting.
S2 PORT	Default: *7000	This is the port the telephone uses to access this system.
S2 ACTION	Default: 1	
S2 RETRY COUNT	<digits between 0 and 255>	Set this to the number of times you want the telephone to retry the connection to the system.
VLAN	0: No VLAN	Choose 0: NO VLAN if no VLAN exists on the network.
	1: Manual VLAN 2: Automatically discover VLAN using DHCP	If DHCP does not exist on the network, or if a remote server supplies DHCP, select number 1 and enter the VLAN ID (see Note 1). If you have the system DHCP active on your system, select 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 your system uses VLAN, check with your network administrator. If your system uses VLAN, the system administrator responsible for the switch can provide the VLAN IDs for your system (see Note 1).
Cfg XAS?	0: No (default) 1: Yes	If you want to enable connection to a Net6 service provider server, choose 1. The system prompts you for an IP address for the server.
Note 1: 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 2 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.

Table 2 IP telephony display messages

Message	Description/Solution
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.

1100 series IP phones

BCM450 supports 1100 series IP phones.T The IP Phone 1110, 1120E, and 1140E. The three phones in the series have a graphical high-resolution LCD display, backlit, with adjustable contrast.

- IP Phone 1110has three user-defined feature keys and four soft keys.
- IP Phone 1120Ehas 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 1140Ehas 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 for 1100 series IP phones

Compatible with the IP Phone 1120E and 1140E, the expansion module includes 18 self-labelling keys. You can connect up to three modules to a phone for a maximum of 54 additional line or feature keys.

1200 series IP phones

The IP Phones 1210, 1220, and 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.

The keys on either side of the navigation keys are programmable keys. The IP Phone 1210 has two keys, while the IP Phone 1220 and 1230 have six keys. The system administrator programs these keys.

Expansion module for 1200 series IP phones

Two expansion module models exist for the IP Phone 1200 series of phones:

- The Expansion Module for IP Phone 1200 series with display.
- 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 or programmable feature keys for your IP Phone. An IP Phone supports up to seven Expansion Modules for IP Phone 1200 Series with display or up to two Expansion Modules for IP Phone 1200 Series with paper labels. An IP phone does not support two different expansion module types on the same phone.

2000 series IP phones

BCM450 supports

IP Phone 2001: connects through an IP link to the BCM450 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 BCM450 line.

IP Phone 2002: connects through an IP link to the BCM450 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 BCM450 line.

IP Phone 2004: connects through an IP link to the BCM450 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 BCM450 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

The IP Key Expansion Module (KEM) can have up to 24 programmable keys (with labels) for IP Phone 2002 or 2004 models with a maximum of four IP KEMs for one phone.

IP Softphone 2050

The IP Softphone 2050 provides VoIP services using a telephony server and your company local area network (LAN). The IP Softphone 2050 includes one-click direct dialing from various windows and applications, twelve user defined feature keys, and four soft keys.

IP Audio Conference Phone 2033

The IP Audio Conference Phone 2033 provides audio conferencing with a keypad that provides many of the set features of the basic Business Series telephones without display or memory buttons. The audio conference phone includes three microphones, and installation instructions.

Analog devices

BCM450 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+, or GASM) for analog devices (see Analog station media bay module (page 22)). To connect a standard analog voice device or data communication device to the BCM system through a digital station module, you must install an ATA2 (see Analog terminal adapter (page 23)).

Analog station media bay module

You can connect a maximum of eight analog telecommunication devices to the analog station media bay modules (ASM8, ASM8+, and GASM). Analog devices include 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:

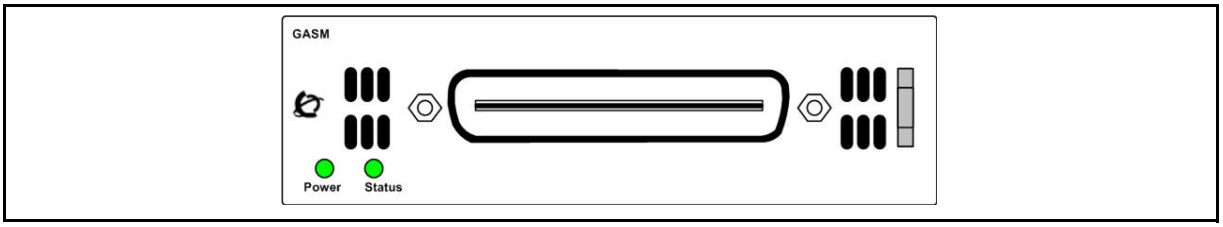
- Visual Message Waiting Indicator (VMWI) LED indicates to the end user that a message is waiting.
- Disconnect supervision (Open Switch Interval [OSI] according to EIA/TIA 464) indicates to the attached device, in an established communication, that the connected device must release the call.

Attention: When disconnect occurs from the central office (CO), the ASM8+ provides an OSI 850 ms (TIA/EIA 464 section 5.4.10.2.4; minimum is 600 ms) to the off-hook station of 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 (TIA/EIA 464 section 5.4.10.2.5[1]). After the station equipment goes on-hook, the ASM8+ station interface restores 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 is designated as an on-premise station (OPS) port.

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

Attention: 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.

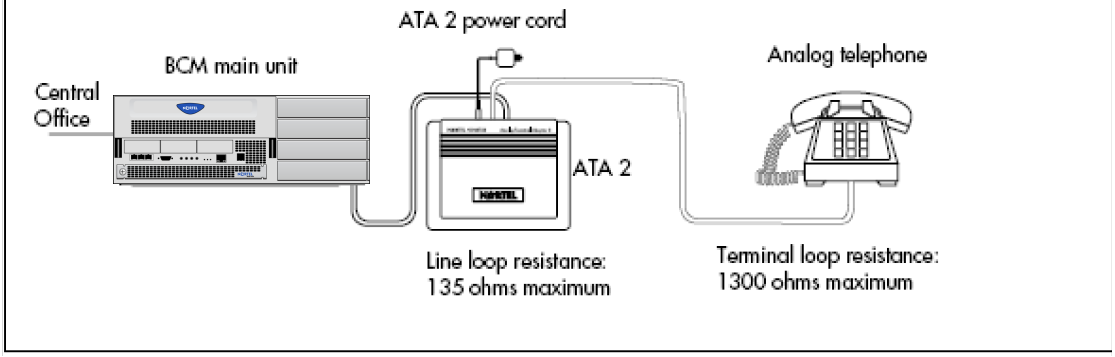
Analog terminal adapter

The analog terminal adapter 2 (ATA2) or ATA connects a standard analog voice device or data communication device to the BCM450 system through a digital station module. Examples of analog voice devices include analog telephones and answering machines. Examples of analog data communication devices include modems and fax machines.

The ATA2 provides on-premise service only (protected plan wiring only).

The following figure shows an installation overview for connecting an analog device or analog data device through an ATA2 to the BCM450 main unit.

Figure 2 Analog device installation overview



Device compatibility and installation requirements

This section provides general information about using Integrated Services Digital Network (ISDN) lines on your Business Communications Manager 450 (BCM) system. You can access detailed information about ISDN through the internet. Your service provider can also provide you with specific information to help you understand what suits your requirements.

Navigation

Release compatibility

Refer to the following topics for release compatibility information:

Release compatibility navigation

ISDN basics

ISDN technology provides a fast, accurate, and reliable means to send and receive voice, data, images, text, and other information through the telecom network.

ISDN uses existing analog telephone wires to multiplex data into separate digital channels, which increases bandwidth.

ISDN uses a single transport to carry multiple information types. Where you once required separate networks for voice, data, images, or video conferencing, it now combines into one common high-speed transport.

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. Digitalization allows ISDN lines to provide higher 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 that operates at 14.4 kbyte/s takes 4.5 minutes to transfer a 1MB data file and a 28.8K modem takes about half that time. If you use one channel of an ISDN line, the transfer time reduces to 1 minute; if you use two ISDN channels, transfer time reduces to 30 seconds.

When you transmit data, the connect time for an average ISDN call is three seconds per call, compared to 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 consists of separate channels, known as B and D channels, that transmit information simultaneously.

- BRI: known as 2B+D because it consists of two B-channels and one D-channel.
- PRI: known as 23B+D (in North America) or 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 bearer channels used to carry voice or data information at speeds of 64 kbyte/s. As each ISDN link (BRI or PRI) includes 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 transmits over a dedicated data channel called the D-channel. The D-channel carries call setup and feature activation information to the destination and includes speeds of 16 kbyte/s

(BRI) and 64 kbyte/s PRI. Data information consists of control and signal information. For BRI only, data information also consists of packet switched data, such as credit card verification.

ISDN layers

ISDN layers refer to the standards established to guide the manufacturers of ISDN equipment, based on the Open Systems Interconnection (OSI) 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. The following three layers work in ISDN for BCM450; to support ISDN service, all three layers must work properly:

- Layer 1: A physical connection that supports fundamental signaling passed between the ISDN network (your service provider) and the BCM450 system. When the LED on a BRI S/T media bay module configured as BRI becomes lit, Layer 1 is functioning.
- Layer 2: A logical connection between the central office or the far end and the BCM450 system. Without Layer 2, call processing is not possible.
- Layer 3: A logical connection between the ISDN network (your service provider) and the BCM450 system. For BRI lines, call processing and service profile identifier (SPID) information exchanges in Layer 3. This controls which central office services are available to the connection. For example, you can program a network connection to carry data calls.

Attention: Service profile identifiers (SPIDs) are a part of the BRI National ISDN standard. SPIDs are not used in the ETSI BRI standard or on PRI.

ISDN bearer capability

Bearer capability describes the transmission standard used by the BRI or PRI line so 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 kbyte/s, restricted 64 kbyte/s or 56 kbyte/s).

Services and features for ISDN BRI and PRI

As part of an ISDN digital network, your system supports enhanced capabilities and features, that include:

- fast 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

PRI services and features

The services and features provided over PRI lines include

- Call-by-call service selection (NI protocol)
- Emergency 911 dialing and 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 kbyte/s per loop (depending on the bandwidth supported by your service provider)
- shared digital lines for voice and data ISDN terminal equipment

BCM450 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. If you connect a POSTA, it 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 include a D-packet handler. To convert the protocol used by the transaction equipment to the X.25 protocol, your ISDN network must also include an integrated X.25 PAD which works with the following versions of the X.25 network: 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 can 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

BCM450 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 BCM450 sets and ISDN terminal equipment with the capability to show the information).
- Multiline hunt or DN hunting that switches a call to another ISDN line if the line usually used by the Network DN is busy (BRI only).
- Sub addressing of terminal equipment (TE) on the same BRI loop. However, terminal equipment which supports sub addressing is not commonly available in North America (BRI only).

BCM450 does not support the transmission of B-channel packet data using nailed up trunks.

Contact your ISDN service provider for more information about these services and features. For more information about ordering ISDN service in North America, see Order ISDN PRI (page 35)and Order ISDN BRI (page 36).

The terminal equipment connected to the BCM450 system can use 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 involved in a call on a public or private network.

Your BCM450 system displays the name of an incoming call when available from the service provider. If the Calling Party Name includes the status of private, it appears as “Private name” (depending on service provider configuration). 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 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 is programmed.

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. You can use name and number blocking with a BCM450 set.

Attention: 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

You can dynamically allocate PRI lines to different service types with the Call by Call feature. You do not have to pre-allocate PRI lines to a given service type. Outgoing calls are routed through a dedicated PRI Pool and the calls can be routed based on various schedules.

Attention: Call by Call service selection for PRI is only available in North America.

Possible service types (depending on your service provider):

- Public: Public service calls connect your BCM450 set with a Central Office (CO). DID and DOD calls are supported.
- Private: Private service calls connect your BCM450 set with a Virtual Private Network (VPN). DID and DOD calls are supported. You can use a private dialing plan.
- TIE: TIE services are private incoming and outgoing services that connect Private Branch Exchanges (PBX), such as BCM450.
- FX (Foreign Exchange): FX service calls logically connect your BCM450 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 that originate 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 can transmit the telephone number and internal extension number of a calling station that dials 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:

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

- All PSTN connections must be over PRI.
- For all sets to be reached from a Public Safety Answering Position (PSAP), you must configure the system for DID access to all sets. In order to reduce confusion, you must configure the dial digits for each set to correspond to the set extension number.
- The OLI digits for each set must be identical to the DID dialed digits for the set.
- The routing table must 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 that includes an appearance on the attendant console.
- The 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 Nortel recommends the person in charge of the system maintain a site map or location directory that allows emergency personnel to rapidly locate a BCM450 set using its DID number. This list should be kept up to date and readily available.

Attention: 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 direct inward dialing

With PRI, you can use the same lines 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 route. Consult your customer service representative to determine whether or not this feature is compatible with your service provider.

Dialing plan and PRI

The dialing plan supports PRI connectivity to public and private networks. The dialing plan is a collection of features responsible for processing and routing incoming and outgoing calls. All PRI calls must go through a dialing plan.

The dialing plan:

- allows incoming calls to route to sets based on service type and digits received
- can 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 BCM450 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 BCM450 main units through the BRI ports (RJ-45) on the front of the main units.

For information about adding integrated lines on an integrated router, see the documentation for your integrated router.

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 BCM450 system connects back-to-back with another switch using a PRI link. If the other switch is loop-timed to your BCM450 system, your DTM (PRI) can be designated as a timing master.

If your BCM450 includes 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

You can program the loops on the BRI module 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. You can individually program each loop 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

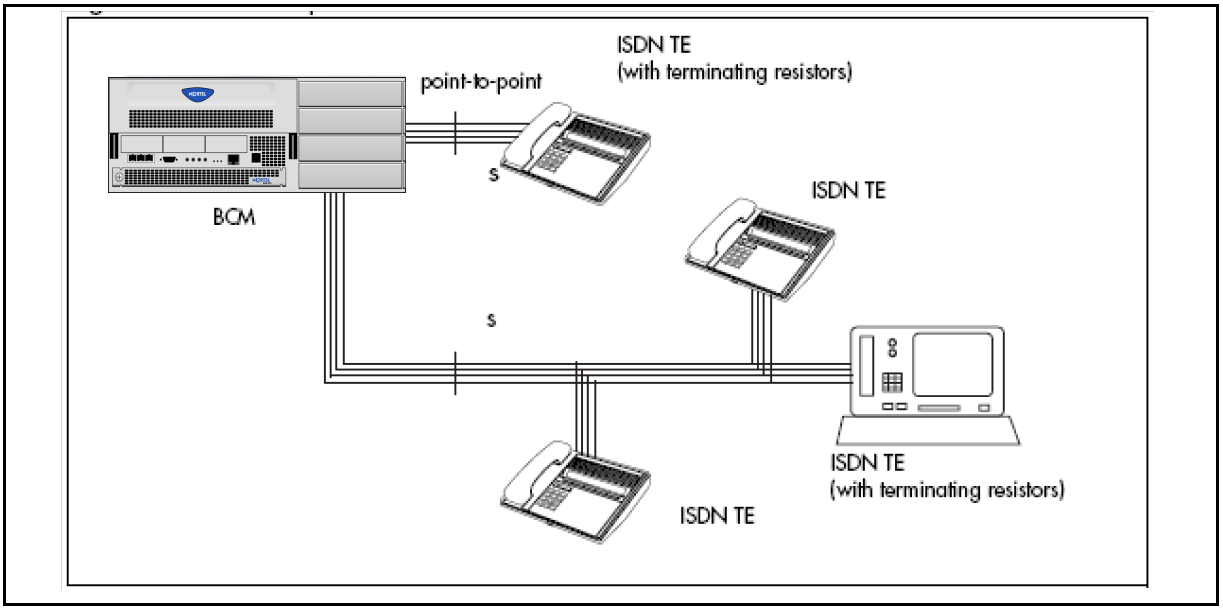
You can add integrated BRI lines on the BCM450 main unit.

S reference point

The S reference point connection provides either a point-to-point or point-to-multipoint digital connection between BCM450 and ISDN terminal equipment (TE) that uses an S interface, see Figure 3 S reference point.

S loops support up to seven ISDN DNs, which identify TE to the BCM450 system.

Figure 3 S reference point

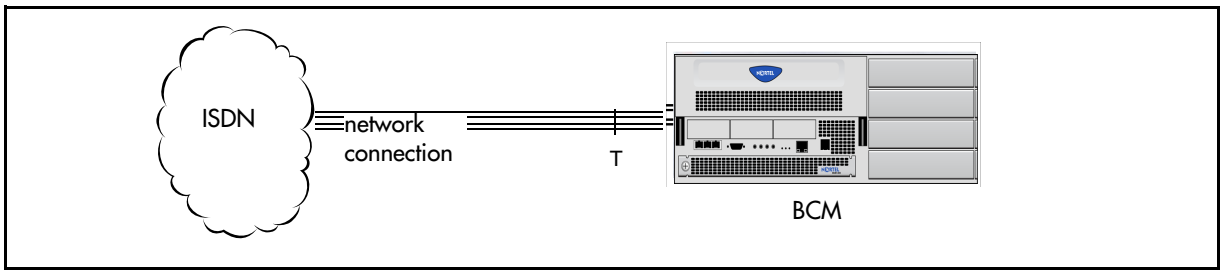


T reference point

The T reference point connections provide a point-to-point digital connection between the ISDN network and BCM450, see Figure 4 T reference point.

T loop provides lines that you can share between all BCM 450 telephones, peripherals and applications, and ISDN TE.

Figure 4 T reference point



You can use A T loop 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 kbyte/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 Integrated Services Digital Network

Systems with ISDN interfaces must 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 BCM450 derives timing from the network using T reference points (loops). Terminal equipment on S reference points (loops) derives timing from the BCM450 system.

When you configure the network connections to the BCM450, 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.
- The system skips S and analog loops, when selecting a network connection for synchronization.

Systems with S loops only 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 includes both a BRI S/T configured as BRI, and a DTM configured as PRI, Nortel recommends that you use PRI as the primary clock source. See PRI hardware (page 32).

ISDN BRI Interface Network Termination Type 1 equipment

The network termination type 1 (NT1) 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 use the U interface.

The NT1 converts and reformats data, so it can transmit to and from the S or T connection (only in North America). In addition, it manages the maintenance messages that travel between the network and the NT1, and between the NT1 and the BCM450 system.

The NT1 from Nortel is packaged two ways:

- a stand alone package that contains one NT1 card (NTBX80XX) and a power supply (NTBX81XX)
- a modular package that contains up to 12 NT1 cards (NTBX83XX) and a power supply (NTBX86AA)

ISDN standards compatibility

In North America, BCM450 ISDN equipment supports National ISDN standards for basic call and calling line identification services. BCM450 BRI is compliant with National ISDN-1 and PRI is compliant with National ISDN-2.

BCM450 does not support Electronic Key Telephone System (EKTS) on PRI.

In Europe, BCM450 supports ETSI Euro and ETSI QSIG standards, and PRI SL-1 protocol.

Plan your Integrated Services Digital 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 BCM450, TE, and other ISDN equipment.

BCM450 does not support packages 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.

Order ISDN PRI

This section provides information about how to order ISDN PRI service for your BCM450.

Order ISDN PRI service in Canada or the United States from your service provider. Set the BCM450 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 or BRI service from your service provider. Set the BCM450 equipment to the Euro ISDN protocol.

Order ISDN BRI

The following sections provide information about how to order ISDN BRI service for your BCM450.

Order 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, that include Call Forwarding.

When you order Microlink service, you must order it with EKTS turned off. If you are using apoint-of-sale terminal adapter (POSTA), ask for D-packet service to be enabled.

Order 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 turned off. Use the following packages as a guideline to order your National ISDN BRI-NI-2. Nortel recommends that you use packages M or P with the BCM450 system. Contact your service provider for more information about the capability packages it offers. Bellcore and National ISDN Users Forum (NIUF) ISDN packages supported by BCM450 (for ordering in U.S.).

Table 3 BCM450 supported ISDN packages

	Capability	Feature set	Optional features	Point-of-sale	Voice	Data
M	Alternate voice or circuit-switched data on both B-channels	--	CLID	--	X	X
P	Alternate voice or circuit-switched data on both B- channels D-channel packet	flexible calling for voice (not supported by BCM450) Basic D-Channel Packet	additional call offering (not supported by BCM450) 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, BCM450 does not support flexible calling for voice and additional call offering features included in package P.

You can order Multi-Line Hunt 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. BCM450 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.

Order ISDN BRI service outside Canada or the United States

Outside of Canada or the United States, order Euro ISDN PRI or BRI service from your service provider. Set the BCM450 equipment to the Euro ISDN protocol.

Supported ISDN protocols

The switch used by your service provider must be running the appropriate protocol 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.

MBM trunk requirements

The following table shows the types and number of lines supported by different MBMs.

Table 4 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
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	

MBM station requirements

The following table shows the types and number of extensions supported by different MBMs.

Table 5 MBM station requirements

Type of extension	Type of MBM	Number of extensions per MBM
Digital extensions	DSM16/DSM16+	16
Digital extensions	DSM32/DSM32+	32
Digital extensions	4x16	16
Digital extensions	G4x16	16
Digital extensions	G8x16	16

Table 5 MBM station requirements

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

IP phones and IP telephony

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 comprise the BCM450 IP telephony and Voice over IP (VoIP) features.

Navigation

IP phone Basics

BCM450 with VoIP provides the following 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 or your 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 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. IP telephone functionality can be transferred between IP telephones using the Hot desking feature. Your telephone features and setup can travel with you between offices.

- Simplicity and consistency: A common approach to service deployment allows further cost-savings by the use of common management tools, resource directories, flow-through provisioning, and a consistent approach to network security. Customers can centrally manage a host of multimedia services and business-building applications through a Web-based browser. The ability to network existing PBXs using IP can bring new benefits to your business. For example, consolidating 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 almost any business system through an analog line, Digital Subscriber Line (DSL), 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 through 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 BCM450 system: IP telephones and Voice over IP (VoIP) trunks. These applications can be used separately or together as a network voice/data solution.

IP telephones

IP telephones offer the functionality of regular telephones, but do not require a hardwire connection to the BCM450. Instead, they must be plugged into an IP network which is connected through the integrated interface (LAN card) on the BCM450.

Calls made from IP telephones through the BCM450 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 (IP Phone Series) or are accessed through your desktop or laptop computer (IP Software Phone 2050).

Voice over IP trunks

VoIP trunks allow voice signals to travel across IP networks. A gateway within the BCM450 converts the voice signal into IP packets, which then transmit 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 or Proxy) and the combined routing table.

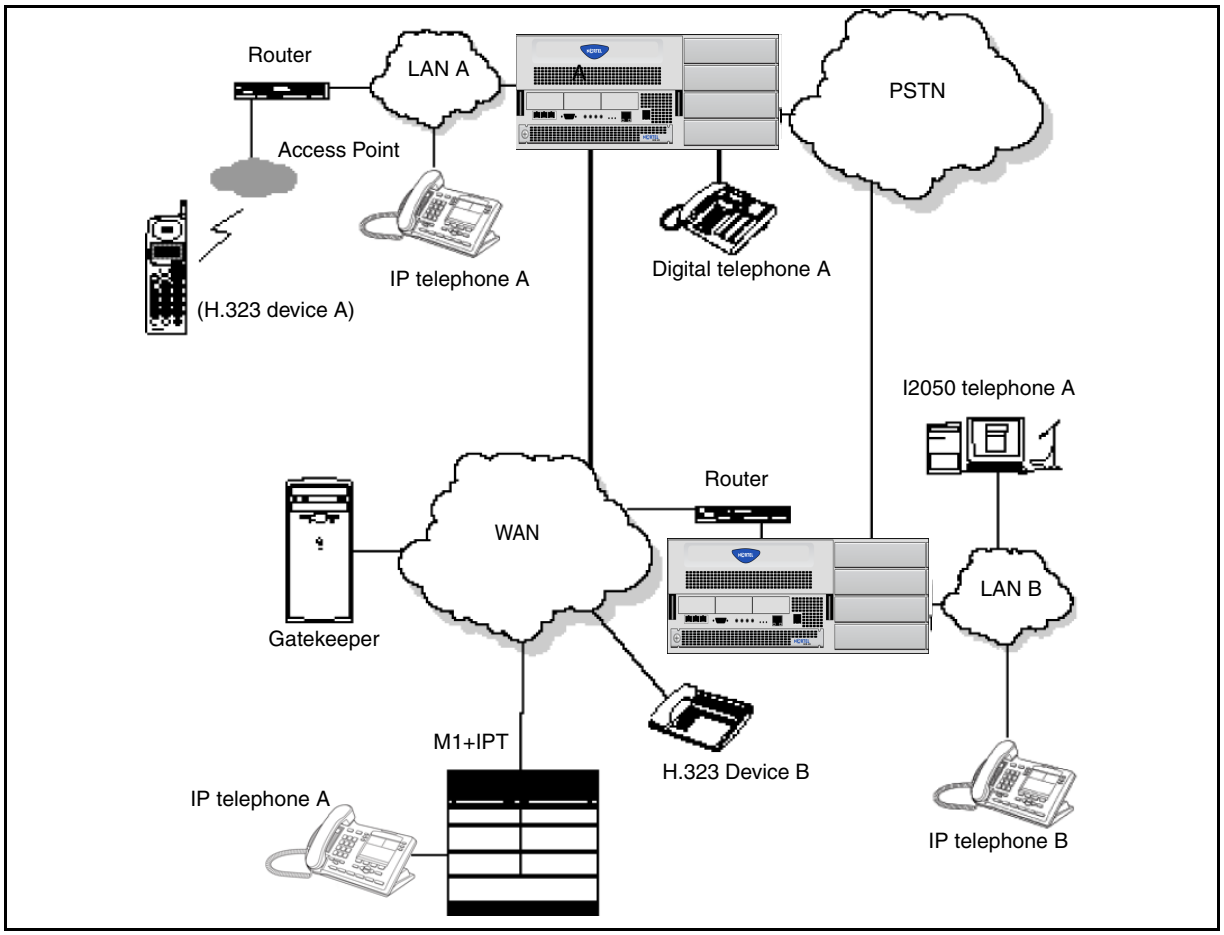
IP telephony networking

The following sections explain the components of the BCM450 system and the devices it interoperates with to create a network.

The following network diagram shows components of a BCM450 network configuration.

In the diagram, two BCM450 systems connect through both a PSTN connection and 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 BCM450 systems use VoIP trunks through a common IP network to connect to the Meridian (M1-IPT) system.

Figure 5 Network diagram



BCM networking

The BCM450 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 BCM450 system connects to devices through multiple IP networks, and through the PSTN. Multiple BCM450 systems can link together on a network of VoIP trunks or dedicated physical lines.

The BCM450 can connect to a LAN through the integrated interface LAN card and to a PSTN through trunk media bay modules, see see Figure 5 Network diagram (page 44). Through these networks, the system accesses other systems and network equipment connected to the network.

Meridian 1 Internet Telephony Path

The Meridian 1 Internet Telephony Path (M1-IPT) allows Meridian 1 systems to communicate with the BCM450 through 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 BCM450s through ISDN PRI SL-1 lines, which provide the same Meridian Customer Defined Network (MCDN) capability that you can achieve through the H.323 VoIP trunks with MCDN active.

A BCM450 connected to an M1-IPT that uses the MCDN protocol can provide access to central voice mail and call attendant systems, which can streamline multi-office telephony administration.

Telephone interoperability

The BCM450 can communicate using digital telephones (Model T7000, T7100, T7208, T7316, T7316E and t7316E+KIMs, and T7406 and T7406E [cordless telephones]), Norstar M-series telephones, ISDN telephones, analog telephones, and IP telephones and applications. With this flexibility, the BCM450 can provide the type of service you require to be most productive in your business.

While analog and digital telephones cannot connect to the BCM450 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 BCM450 to voice channels.

The IP telephones connect to the BCM450 across an IP network through either a LAN or a WAN. From the BCM450 connection, they can use standard lines or VoIP trunks to communicate to other telephones on other public or private networks. The BCM450 also supports H.323 (version 4) and H.323 third-party devices through this type of connection.

Network gatekeepers

A gatekeeper tracks the 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 BCM450 system attaches, but gatekeepers can be useful on networks with a large number of devices, see see Figure 5 Network diagram (page 44).

For example: Digital telephone A wants to call IP telephone B, which is attached to BCM450 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 BCM450 B over the network. BCM450 B then passes the call to IP telephone B.

The BCM450 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 BCM450 is compatible with Nortel Communication Server 1000 (CSE1K) gatekeepers.

CAUTION Meridian 1 IPT does not support the RadVision gatekeeper.

IP network

see Figure 5 Network diagram (page 44) shows several LANs and a WAN in the network. When you plan your network, consider all requirements for a data network. Your network administrator should be able to advise you about the network setup and how the BCM450 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 BCM450, a WAN is any IP network connected to a WAN card on the BCM450 system. This may also be a direct connection to another BCM450 system.

If you want to deploy IP telephones that will connect to a LAN outside of the LAN that the BCM450 is installed on, you must ensure the BCM450 can 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 BCM450, a LAN is any IP network connected to the integrated interface (a LAN card) on the BCM450 system. Often, the LAN can include a router that forms a connection to the Internet. A BCM450 can include 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 BCM450 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 become 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.

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 (kbit/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 usage, you can achieve better voice quality by configuring the IP telephone to use a G.729 Codec.

You can only change the codec on a configured IP telephone if it is online to the BCM450, or if Keep DN Alive is enabled for an offline telephone.

The BCM450 supports the following codecs:

- G.729
- G.729 with Voice Activity Detection (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.

Attention: You can only change the jitter buffer on a configured IP telephone if it is online to the BCM450, or if Keep DN Alive is enabled for an offline telephone.

The jitter buffer works in the following manner (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 to ensure a constant rate of arriving frames at the handset speaker.

The delay of packets can provide 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 (two jitter-sized) 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 BCM450 system IP telephones include:

- 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)

Quality of service routing

To minimize voice jitter and maximize quality of service (QoS) over low bandwidth connections, the BCM450 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 separate for Per Hop Behavior (PHB). The DSCP is contained within the DiffServ field, which was formerly known as the ToS field. The BCM450 assigns Expedited Forwarding (EF) PHB for voice media packets. On the BCM450, these assignments cannot be adjusted.

Telephone relocation

This following sections explain how you can physically move a telephone within the system so that the telephone programming follows the telephone to the new location.

Digital telephone relocation

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.

The set relocation feature applies to digital and analog telephones only. IP telephones always retain their programming. For more information, see IP telephone relocation without changing the DN (page 80) or IP telephone relocation with a changed DN (page 80).

Digital telephone relocation recommendations

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 complete. Moving the telephone again before the three-minute period is up can result in loss of programming.

IP telephones relocation

IP telephones retain their DN when you move them to a new location on the same subnet.

In some circumstances, you may want your IP telephone to 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.

Media bay module configuration

You can install media bay modules (MBM) in BCM450 main units and expansion units, depending on your system requirements.

Prerequisites

- Ensure your system includes an installed MBM with assigned DS30 resources and configured dip switches.
- For more information about installing an MBM, see Nortel Business Communications Manager 450 1.0 Installation—System Guide (NN40160-301).
- An experienced installer can wire the analog station module (ASM) for your system using the wiring chart, for more information see the Nortel Business Communications Manager 450 1.0 Installation—System(NN40160-301).
- 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.

Navigation

- Configuring resources for the MBM (page 51)

Configuring resources for the MBM

This procedure describes how to configure your MBM.

Step	Action
1	Open **Element Manager** and connect to your BCM450 system.
2	Click **Configuration**, **Resources**, **Telephony Resources.** The Telephony Resources panel appears.
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 want to install, double-click the cell of the row you selected to reveal a list and select the MBM type.

5 Click **Configure** and enter the required information in the Configure box.

6 Click **Ok.**

7 Repeat steps step 4 through step 5 to enable each MBM in your system.

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

--End--

IP phone registration

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

Registering Nortel 11XX, 12xx, and 20XX IP telephones on the system includes two steps; setting up registration in Element Manager and configuring each IP phone. For more information about configuring your IP phone see, IP phone configuration (page 16).

Prerequisites

- Ensure you loaded the appropriate keycodes to activate the Nortel IP telephones on your BCM450 system.

Navigation

Enabling registration in Element Manager

Use Element Manager to set up the system programming to receive registrations under the Telephony Resources panel.

Prerequisites

- Start Element Manager.

Procedure steps

Step	Action
1	In **Element Manager** navigate to **Resources**, **Telephony Resources**.
2	In the Telephony Resources table, click the **IP Sets** row. The Details for Module panel appear below the Telephony Resources table.
3	In the **IP Terminal Global Settings** tab, select the **Enable registration** check box.
4	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.
5	If you want the system to automatically assign DN records to the telephones, select the **Auto-assign DNs** check box.

--End--

Automatically assign directory names

You can automatically configure IP Phones with directory names (DNs) assigned.

Prerequisites

- Start Element Manager.

Procedure steps

Step	Action
1	In Element Manager navigate to **Resources**, **Telephony Resources**.
2	In the **Telephony Resources** table, click the **IP Sets** row. The Details for Module panel appear below the Telephony Resources table.
3	Select the **Enable registration** check box.
4	Select the **Enable global registration password** check box.
5	Leave the **Global password** field blank.
6	Select the **Auto-assign DNs** check box.
7	Clear the **Enable registration** check box after the IP Phones become operational. Turn Enable registration and Auto-assign DNs off after the telephones register. Nortel cautions that leaving your IP registration open and unprotected by a password can pose a security risk.

--End--

Registering IP phones in the system

To register your IP phone in the system, perform the following procedure.

Procedure steps

Step	Action
1	Connect the telephone to the IP connection. If a password is entered in the Terminal Registration screen and the telephone is not yet registered, the telephone prompts you for that password.
2	If prompted for a password, type the password.

Attention: If Auto Assign DN is not selected, the telephone prompts you for a DN. See Nortel Business Communications Manager 450 1.0 Configuration—Devices(NN40020-300) for more information about configuring telephones.

3	If prompted for a DN, type the DN.
4	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.

--End--

Accessing the local configuration menu on an IP Phone 2001, 2002 or 2004

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.

Procedure steps

Step	Action

1 Restart the telephone by disconnecting the power.

2 Reconnect the power.

After four seconds, the top light flashes and “Nortel Networks” appears on the screen.

3 When the greeting appears, immediately press the four display buttons one at a time, from left to right.

These buttons are located directly under the display.

--End--

Accessing the local configuration menu on an IP Phone 2033

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.

Procedure steps

Step	Action
1	Disconnect the power.
2	Reconnect the power. After four seconds, the top light flashes and “Nortel Networks” appears on the screen.
3	When the greeting appears, immediately press the three display buttons one at a time, from left to right. These buttons are located directly under the display.

--End--

Accessing the local configuration menu on an IP Phone 2007

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.

Procedure steps

Step	Action
1	Disconnect the power.

2	Reconnecting the power. After four seconds, “Nortel Networks” appears on the screen.
3	When the greeting appears, immediately press 007* on the dialpad.

--End--

Accessing configuration menu on an IP Phone 1120E or 1140E

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.

Procedure steps

Step	Action
1	Press the **Services** key twice to access the Local Tools menu.
2	Use the navigation keys to the find the service to configure.
3	Press **Select**. Press the button sequence with 1.5 seconds or the telephone does not enter configuration mode. If Manual CfgDHCP (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 address.
5	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.

--End--

Accessing configuration menu on an IP Phone 1210, 1220, or 1230

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.

Many of the options discussed in this section are for system administrator use only. Do not make changes unless you are instructed by your system administrator.

Procedure steps

Step	Action
1	Press the **Services** key twice to access the Local Tools menu. If you attempt to access the Local Tools menu and a dialog box appears that prompts you for a password, contact your system administrator.
2	From the Local Tools menu, press the key on the dialpad to open the chosen menu.
3	From the selected menu, press the key on the dialpad to open the chosen submenu.
4	Use the navigation keys and **Select** key to make a selection.
5	Press the **Cancel** soft key to exit the menu or submenu.

--End--

Deregistering IP telephones

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

Attention: When you deactivate the phone all active calls drop.

Procedure steps

Step	Action
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.

--End--

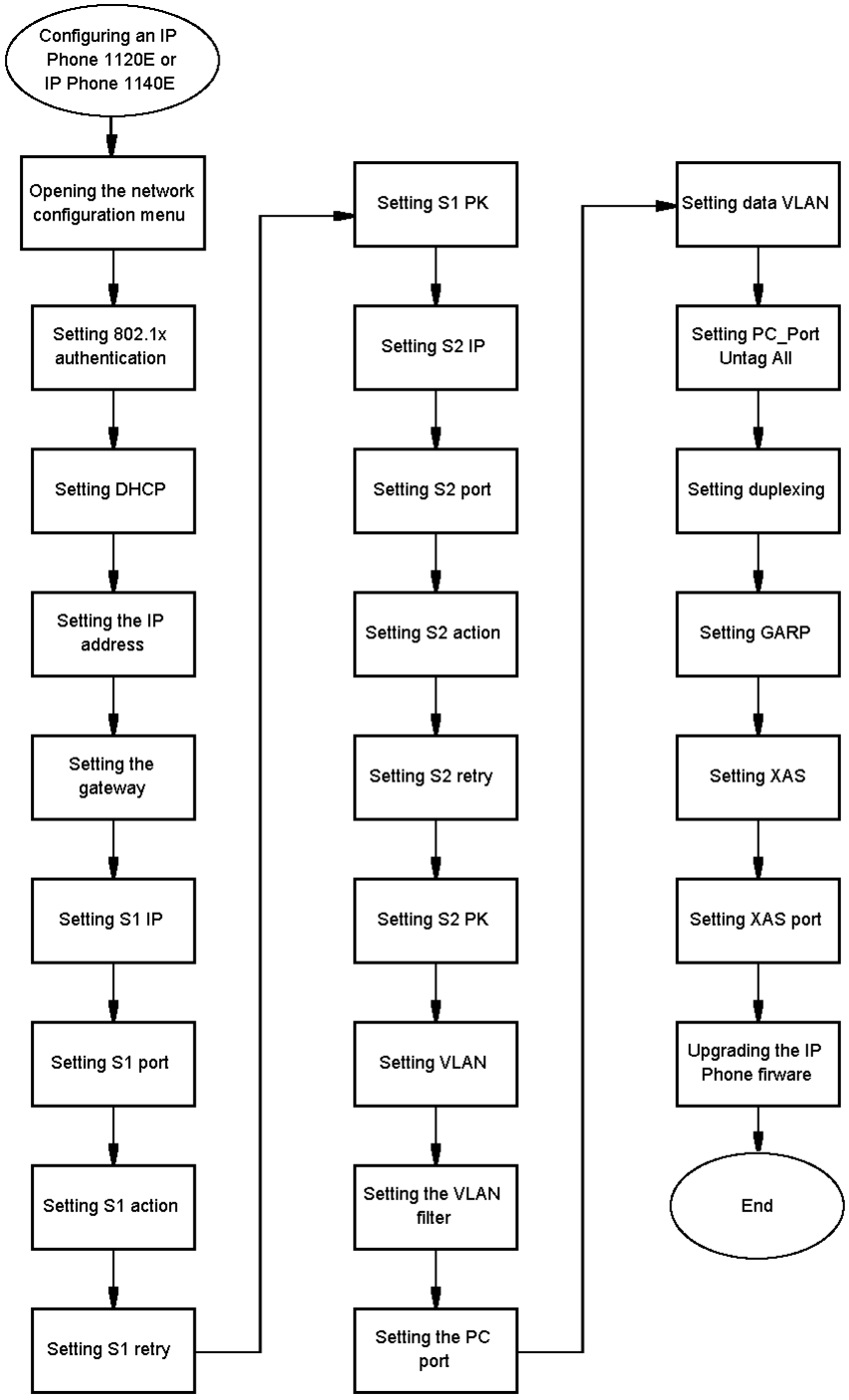
IP Phone 1120E or IP Phone 1140E configuration

When you configure an IP Phone 1120E or IP Phone 1140E, you have 45 seconds between plugging in the IP Phone 1120E power adapter and the appearance of the "Nortel" text. When you see the "Nortel" text, you have one second to respond by pressing each soft key at the bottom of the display in sequence from left to right. If you miss the one second response time, the IP Phone 1120E attempts to locate and to connect to the server. You can begin the power-up sequence again, or double-press the Services key to open the Local diagnostic utilities to access the IP Phone settings.

IP Phone 1120E or IP Phone 1140E configuration procedures

This task flow shows the sequence of procedures you perform to configure the IP Phone 1120E or IP Phone 1140E. To link to any procedure, click on IP Phone 1120E or IP Phone 1140E configuration navigation (page 63).

Figure 6 IP Phone 1120E or IP Phone 1140E configuration



IP Phone 1120E or IP Phone 1140E configuration navigation

Opening the network configuration menu

To begin to configure your phone, access the menu.

Procedure steps

Step	Action
1	Restart the telephone by disconnecting and reconnecting the power.

A splash screen appears with the Nortel logo on it. It includes a display interval of 2 to 3 seconds with a 10 second pause.

There is approximately 45 seconds between plugging in the IP Phone 1120E or 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.

The Network Configuration menu opens.You can press the Apply&Reset soft key to save settings and reset the IP Phone.You can press the Exit soft key to exit the menu without saving changes and return to the Network Configuration menu.

--End--

Setting 802.1x authentication

Your IP Phone 1120E and1140E firmware supports 802.1x authentication. The authentication is dependent on your layer 2 switches and is not part of the BCM450. You can choose to enable 802.1x on your IP 1120E or 1140E phone.

Prerequisites

- Open the network configuration menu, see Opening the network configuration menu (page 63).
- Ensure when the Network Configuration menu opens that the **Enable 802.1x** check box is selected (press **Enter** to toggle the item).

Procedure steps

Step	Action
1	Press **Enter** to start Edit mode.
2	Use the Right navigation key to scroll and select the **Enable 802.1x (EAP)** check box.
3	Use the keypad to supply the following information: • Device ID • Password (26567*738) • Retype password
4	Press **Enter**.
5	Use the Right navigation key to scroll and select the **Enable 802.1ab** (LLDP Enable) check box.
6	Press **Enter** to toggle this item on and off.

--End--

Setting DHCP

You can use DHCP with your phone.

Prerequisites

- For first time IP Phone configuration, complete Setting 802.1x authentication (page 64).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select DHCP command.
2	Press **Enter**.
3	Press the **Down** navigation key to open list box.

Attention: You must also install, configure, and run a DHCP server and DHCP relay agent if you choose Partial DHCP, or Full DHCP configuration.

4 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 Private Key (PK) numbers manually.
- **Full**—IP network information, Server 1 IP address, Server 2 IP address, and XAS information are provided by the DHCP server.

5 Press **Enter**.

--End--

Setting the IP address

Assign an IP address to your phone before you use it.

Prerequisites

- For first time IP Phone configuration, complete Setting DHCP (page 65).

- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Set IP**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Set IP** information (a valid IP Phone 1120E IP address).

--End--

Setting the gateway

Assign a gateway to your phone before you use it.

Prerequisites

- For first time IP Phone configuration, complete Setting the IP address (page 65).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Gateway**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Gateway** information (the default gateway for the IP Phone 1120E or IP Phone 1140E on the LAN segment to which it is connected).

--End--

Setting S1 IP

Assign the primary IP address of your BCM 450 before you use the phone.

Prerequisites

- For the first time IP Phone configuration, complete Setting the gateway (page 66).

- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S1 IP**.
2	Press **Enter** to start the Edit mode.
3	Use the dialpad to enter the **S1 IP** information (the primary IP address of the BCM 450 node).

--End--

Setting S1 port

Assign the port your phone will use.

Prerequisites

- For first time IP Phone configuration, complete Setting S1 IP (page 66).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Port.**
2	Use the dialpad to enter the **S1 Port** information (a fixed value of 7000)

--End--

Setting S1 action

Assign the TPS value before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S1 port (page 67).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S1 Action**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the following information: • for TPS only, enter 1 • for TPS and Secure Media Controller, enter 6 or 1

--End--

Setting S1 retry

Assign the number of connection retries your phone attempts before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S1 action (page 67).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Retry**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Retry** information (the number of times the IP Phone 1120E or IP Phone 1140E attempts to connect to the server).

--End--

Setting S1 private key

Assign the private key of the Secure Media Controller to which the IP Phone is connected before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S1 retry (page 68).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S1 PK**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the information.
4	Press **Enter** to start Edit mode.
5	Use the dialpad to enter the private key (16-digit hexadecimal number, the default is ffffffffffffffff).

Attention: To enter letters, you must hit the "*" key and then the keypad for the corresponding letter. For example, 1=a, 2=b, 3=d.

--End--

Setting S2 IP

Assign the secondary IP address of your BCM 450 before you use the phone. The IP Phone 1120E and IP Phone 1140E can support a primary (S1) and secondary (S2) connect server.

Prerequisites

- For first time IP Phone configuration, complete Setting S1 private key (page 68).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S2 IP**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **S2 IP** information (the secondary BCM 450 node IP address for the IP Phone 1120E or IP Phone 1140E).

Attention: If a secondary connect server is not in use, an IP address is not required.

--End--

Setting S2 port

Assign the port your phone will use.

Prerequisites

- For first time IP Phone configuration, complete Setting S2 IP (page 69).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Port**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Port** information (same as **S1 Port**).

--End--

Setting S2 action

Assign the TPS value before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S2 port (page 70).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S2 Action**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the following information: • for TPS only, enter 1 • for TPS and Secure Media Controller, enter 6 or 1

--End--

Setting S2 retry

Assign the number of connection retries your phone attempts before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S2 action (page 70).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Retry**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Retry** information (same as S1).

--End--

Setting S2 private key

Assign the Private key of the alternate Secure Media Controller to which the IP Phone is connected before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S2 retry (page 71).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **S2 PK**.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the information. Set **S2 PK** to **6** or **1**.
4	Press **Enter** to start edit mode.

5	Use the dialpad to enter the private key (16-digit hexadecimal number, the default is ffffffffffffffff).

Attention: To enter letters, you must hit the "*" key and then the keypad for the corresponding letter. For example, 1=a, 2=b, 3=d.

--End--

Setting VLAN

Your IP Phone 1120E and1140E firmware supports VLAN. The VLANs are dependent on your layer 2 switches and are not part of the BCM450. You can choose to assign the VLAN mode options on your IP 1120E or 1140E phone.

Prerequisites

- For first time IP Phone configuration, complete Setting S2 private key (page 71).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **VoiceVLAN** command.
2	Press **Enter**.
3	Press the **Down** navigation key to open the list box.
4	Use the **Up** or **Down** navigation key to scroll and select 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.
5	Press **Enter**.

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.

--End--

Setting the VLAN filter

Set VLAN filtering options before you use the phone.

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 appears dimmed if you select **No in the VoiceVLAN** command.

Prerequisites

- For first time IP Phone configuration, complete Setting VLAN (page 72).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select the **VLAN filter** check box.
2	Press **Enter** to toggle this item on and off.

--End--

Setting the PC port

Set the PC port options before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting the VLAN filter (page 73).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **Disable PC Port.**
2	Press **Enter** to toggle this item on and off.

--End--

Attention: If Disable PC Port is enabled, Data Vlan, PC-Port Untag All, and Cached IP are dimmed and cannot be enabled.

Setting data VLAN

Assign data mode options before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting the PC port (page 73).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select **DataVLAN** command.
2	Press **Enter**.
3	Press the **Down** navigation key to open the list box.

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

4	Use the **Up** or **Down** navigation key to scroll and select 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.
5	Press **Enter**.

--End--

Setting PC_Port Untag All

Assign the tag option before you use the phone.

If DATA VLAN is enabled, the tag on all traffic destined for the PC port is stripped, by default. To override this action, clear 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.

Prerequisites

- For first time IP Phone configuration, complete Setting data VLAN (page 74).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select the **PC-Port Untag All** check box.
2	Press **Enter** to toggle this item on and off.

--End--

Setting the duplex mode

Assign the duplex mode before you use the phone.

Prerequisites

- For first time IP Phone configuration, complete Setting PC_Port Untag All (page 74).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select the **Duplex** command.
2	Press **Enter**.
3	Press the down navigation key to open the list box and select one of the following options: • **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.

--End--

Setting Gratuitous ARP

Assign Gratuitous ARP (GARP) options before you use the phone. The GARP feature protects the IP Phone from a GARP Spoof attack from the network.

The SRTP media encryption feature provides encrypted media. The IP Phone 1120E and 1140E firmware supports SRTP media encryption. A preshared secret is embedded in the Nortel IP Phone firmware to generate and to exchange encryption parameters without BCM450 involvement.

Prerequisites

- For first time IP Phone configuration, complete Setting the duplex mode (page 75).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select the **Ignore GARP** check box.
2	Press **Enter** to toggle this item on and off.
3	Select **Yes** to enable SRTP media encryption or select **No** to disable media encryption.
4	Use the **Right** navigation key to scroll to **Enable PSK SRTP**.
5	Press **Enter** to toggle this item on and off.

--End--

Setting External Application Server

You can choose to use an external application server (XAS) with your phone. The XAS delivers business applications to the IP Phone.

Prerequisites

- For first time IP Phone configuration, complete Setting Gratuitous ARP (page 76).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	If an External Application Server (XAS) is available in the network, use the right navigation key to scroll and select the **XAS IP** combo box.
2	Use the dialpad to enter the XAS IP address.
3	If the XAS supports graphical displays, use the right navigation key to scroll and select **Graphical XAS** check box.
4	Press **Enter** to toggle this item on and off. If Graphical XAS is not enabled, Port is dimmed out.

--End--

Setting the XAS port

Assign a port for your phone to use the XAS.

Prerequisites

- For first time IP Phone configuration, complete Setting External Application Server (page 76).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Use the **Right** navigation key to scroll and select the **Port** command.
2	Press **Enter** to start Edit mode.
3	Use the dialpad to enter the **Port** information.

--End--

Upgrading the IP phone firmware

You can upgrade the IP Phone 1120E or 1140E firmware two ways:

- Register your IP phone. 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. To register your IP Phone see, Registering IP phones in the system (page 55).
- The IP Phone 1120E or 1140E supports remote firmware upgrades through a TFTP process and an automated UFTP process.

Prerequisites

- For first time IP Phone configuration, complete Setting the XAS port (page 77).
- If the IP Phone is already configured, complete Opening the network configuration menu (page 63).

Procedure steps

Step	Action
1	Enter the TFTP IP address.
2	Enable Bluetooth.

--End--

Telephone relocation

You can physically move a telephone within the system so that the telephone programming follows the telephone to the new location.

Navigation

Digital telephone relocation

To move a digital telephone you must first enable set relocation in Element Manager.

Procedure steps

Step	Action
1	In the Element Manager, go to **Configuration**, **Telephony**, **Global Settings**, **Feature Settings.**
2	In the **Feature Settings** section, select the **Set relocation** check box. 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.
3	Clear the **Set relocation** check box.

--End--

Keeping an IP telephone active

To keep an IP telephone active after it is disconnected, you must change a setting in Element Manager.

Procedure steps

Step	Action
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. Clear the **Keep DN alive** check box to allow the Direct Number (DN) record to become inactive if the IP telephone is disconnected.

--End--

IP telephone relocation without changing the DN

To move an IP telephone without changing the DN, complete this procedure.

Procedure steps

Step	Action
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 or S2 IP address. If your network is using partial DHCP, reconfiguration is not required at this step.
5	Disconnect the power from the IP telephone or three-port switch.

--End--

IP telephone relocation with a changed DN

To move an IP telephone and change the DN, complete this procedure.

Procedure steps

Step	Action

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.

--End--

Media encryption on IP phones

Media encryption is available on Nortel IP phones. The IP Phone firmware delivers the media encryption enhancement, through Secure Realtime Transport Protocol (SRTP) using pre-shared keys (PSK), to Nortel's IP phones without the involvement of the BCM450.

For the most recent documentation about your IP phone, see the Nortel support portal at:
http://support.nortel.com/go/main.jsp?cscat=CATEGORY&catOID=-9604

Prerequisites

- Ensure you loaded the appropriate keycodes to activate the Nortel IP telephones on your BCM450 system.

Navigation

- Configuring SRTP media encryption on a Phase II IP Phone 2001, 2002, and 2004 (page 83)
- Configuring SRTP media encryption on an IP Phone 2007 (page 84)
- Configuring SRTP media encryption on an IP Phone 1110, 1120E, and 1140E (page 85)
- Configuring SRTP media encryption on an IP Phone 1210, 1220, and 1230 (page 85)

Configuring SRTP media encryption on a Phase II IP Phone 2001, 2002, and 2004

Complete the following procedure to enable Secure Real-Time Transport Protocol (STRP) media encryption.

Procedure steps

Step	Action
1	Reset the phone by disconnecting and re-connecting power.
2	When the Nortel logo appears, press each of the four soft keys in sequence.

3 If no other configuration changes are required, press **OK** repeatedly until the "PSK SRTP? (0-No, 1-Yes)" option appears.

4 Press **1** to enable the PreShared Key (PSK) SRTP or **0** to disable PreShared Key (PSK) SRTP.

5 Select **OK** to confirm the change.

6 Restart the IP Phone.

When the IP Phone is restarted, the firmware sets the encryption support accordingly. SRTP Media Encryption is disabled by default on the IP Phones. If enabled, when a secure call is established, the phone displays a small lock icon to indicate the media is encrypted. The icon is in the first position of the first line of the caller display. The icon reflects the status of the main media stream.

--End--

Configuring SRTP media encryption on an IP Phone 2007

Complete the following procedure to enable Secure Real-Time Transport Protocol (STRP) media encryption.

Procedure steps

Step	Action
1	Tap the **Tools** icon.
2	Select **Network Configuration** menu.
3	Scroll to the PSK SRTP combo box. The current setting is displayed.
4	Select **Yes** to enable PSK SRTP or select **No** to disable PSK SRTP.
5	Tap the **Apply&Reset** soft key to save the change and to restart the phone.

When the IP Phone is restarted, the firmware sets the encryption support accordingly. SRTP Media Encryption is disabled by default on the IP Phones. If enabled, when a secure call is established, the phone displays a small lock icon to indicate the media is encrypted. The icon is in the first position of the first line of the caller display. The icon reflects the status of the main media stream.

--End--

Configuring SRTP media encryption on an IP Phone 1110, 1120E, and 1140E

Complete the following procedure to enable Secure Real-Time Transport Protocol (STRP) media encryption.

Procedure steps

Step	Action
1	Double-press the **Services** key.
2	Press **3** on the dialpad to access the Network Configuration menu or use the Up/Down navigation keys to scroll and highlight the Network Configuration option.
3	Press **Enter** (center button of navigation cluster) to start the edit mode.
4	Use the **Right** navigation key to navigate to the **Enable PSK SRTP** combo box. The current setting is displayed.
5	Press **Enter** to toggle the check box to enable (checked) or disable (blank) PSK SRTP.
6	Press the **Apply** soft key to save the change and to restart the phone.

When the IP Phone is restarted, the firmware sets the encryption support accordingly. SRTP Media Encryption is disabled by default on the IP Phones. If enabled, when a secure call is established, the phone displays a small lock icon to indicate the media is encrypted. The icon is in the first position of the first line of the caller display. The icon reflects the status of the main media stream.

--End--

Configuring SRTP media encryption on an IP Phone 1210, 1220, and 1230

Complete the following procedure to enable Secure Real-Time Transport Protocol (STRP) media encryption.

Procedure steps

Step	Action
1	Double-press the **Services** key.
2	Press **3** on the dialpad to access the Network Configuration menu or use the Up/Down navigation keys to scroll and highlight the Network Configuration option.

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

4 Use the **Right** navigation key to navigate to the **Enable PSK SRTP** combo box. The current setting is displayed.

5 Press **Enter** to toggle the check box to enable (checked) or disable (blank) PSK SRTP.

6 Press the **Apply&Reset** soft key to save the change and restart the phone.

When the IP Phone is restarted, the firmware sets the encryption support accordingly. SRTP Media Encryption is disabled by default on the IP Phones. If enabled, when a secure call is established, the phone displays a small lock icon to indicate the media is encrypted. The icon is in the first position of the first line of the caller display. The icon reflects the status of the main media stream.

--End--

Device Compatibility

To determine if your devices are compatible with your BCM, see Table 6 Telephony devices release compatibility matrix (page 87). The matrix displays telephony devices and the BCM releases with which they are compatible.

Table 6 Telephony devices release compatibility matrix

Device	BCM 3.6	BCM 3.7	BCM 4.0	BCM50 1.0	BCM50 2.0	BCM50 3.0	BCM450	MBM
T7000 (EU only)	X	X	X	X	X	X	X	DSM
T7100	X	X	X	X	X	X	X	DSM
T7208	X	X	X	X	X	X	X	DSM
T7316	X	X	X	X	X	X	X	DSM
T7316E	X	X	X	X	X	X	X	DSM
T7406 (North America only)	X	X	X	X	X	X	X	DSM
T7406E						X	X	DSM
T 24 KIM	X	X	X	X	X	X	X	DSM
M7208	X	X	X	X	X	X	X	DSM
M7324	X	X	X	X	X	X	X	DSM
M7406	X	X	X	X	X	X	X	DSM
Central Answering Position (CAP)	X	X	X	X	X	X	X	DSM
BST Doorphone	X	X	X		X	X	X	DSM
Dect 413x		X	X		X	X	X	DSM
Dect 414x		X	X		X	X	X	DSM
Digital Mobility Phone 7420					X	X	X	DSM
Digital Mobility Phone 7430		X			X	X	X	DSM

Table 6 Telephony devices release compatibility matrix

Device	BCM 3.6	BCM 3.7	BCM 4.0	BCM50 1.0	BCM50 2.0	BCM50 3.0	BCM450	MBM
Digital Mobility Phone 7440		X			X	X	X	DSM
IP Phone 1110			X			X	X	
IP Phone 1120E			X		X	X	X	
IP Phone 1140E			X		X	X	X	
Expansion Module IP Phone 1100 Series						X	X	
IP Phone 1210						X	X	
IP Phone 1220						X	X	
IP Phone 1230						X	X	
IP Phone 2001	X	X	X	X	X	X	X	
IP Phone 2002	X	X	X	X	X	X	X	
IP Phone 2004	X	X	X	X	X	X	X	
IP Phone 2007		X	X		X	X	X	
IP KEM			X		X	X	X	
IP Softphone 2050	X	X	X	X	X	X	X	
IP Audio Conference Phone 2033			X		X	X	X	
WLAN 2210 Handset		X	X		X	X	X	
WLAN 2211 Handset		X	X		X	X	X	
WLAN 2212 Handset			X		X	X	X	
WLAN 6120						X	X	
WLAN 6140						X	X	

IP Phones reference

When you register your phone, you must configure it using phone buttons, the Tool icon, or the Services key. The type of phone you register determines which of these methods to use.

Figure 7 IP phone buttons, tool icon and service keys

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